Synway SMG Series Wireless Gateway

SMG4008

Wireless Gateway

User Manual

Version 1.0.0

Synway Information Engineering Co., Ltd

www.synway.net

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Revision History

Version	Date	Comments
Version 1.0.0	2015-08	Initial publication

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Chapter 1 Product Introduction

Thank you for choosing Synway SMG Series Wireless Gateway!

The Synway SMG series wireless gateway products (hereinafter referred to as 'GSM gateway'), as a part of the Synway gateway products, works mainly for connecting the GSM network with the VoIP network. It adopts an updated VoIP processor and the GSM module, uses the push-pull SIM card socket for easy replacement of the SIM card, quite advanced in technology. So far, only SMG4008 is available.

1.1 Typical Application

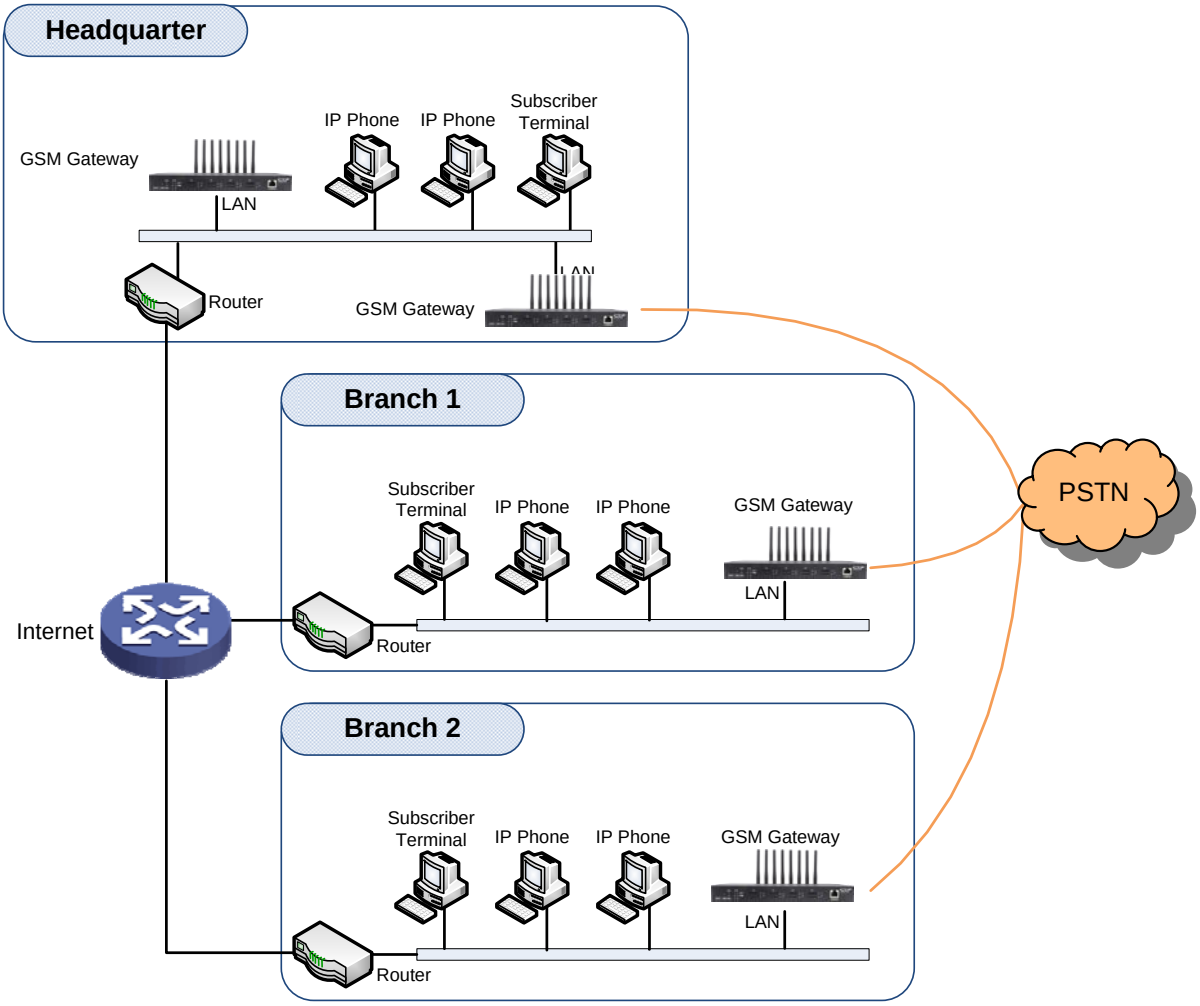

Figure 1-1 Typical Application

1.2 Feature List

Basic Features	Description
TDM Call	Call initiated from TDM to IP, via routing and number manipulation to obtain the called IP address.

IP Call	Call initiated from IP to TDM, via routing and number manipulation to obtain the call destination.
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number.
Call Forward	Three options available: Unconditional, Busy, No Reply and Unreachable.
CID	Displays the CallerID.
Echo Cancellation	Provides the echo cancellation feature for a call conversation over the wireless channel.
TDM/VoIP Routing	Sets a routing path: from IP to TDM or from TDM to IP.
Simultaneous Register to Multiple Servers	Registers the gateway to a master registrar server and a spare registrar server simultaneously.
IMS Network	Registers the gateway to a server under IMS network.
Custom IVR Recording	Provides the interface to customize the IVR Recording.
White/Black List	Allows the setting of the white/black list for WEB access.
Voice Gain Adjust	Supports the gain adjustment for the received or sent voice.
Receive or Send SMS/USSD	Supports the SMS sending and receiving, as well as the USSD request and response.
Auto Select Network	Supports the auto identification and selection of the network operator.
SMS CODEC	Two options available: ASCII and UCS2.
Signaling & Protocol	**Description**
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261.
Voice	CODEC: G.711A, G.711U, G.729A/B, G.723, G.722, AMR, iLBC DTMF Mode: RFC2833, SIP INFO, INBAND
Network	**Description**
Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.
Static IP	IP address modification support.
DHCP	IP address dynamic allocation support.
DNS	Domain Name Service support.
Security	**Description**
Admin Authentication	Supports admin authentication to guarantee the resource and data security.
System Monitor	Monitors the running status of the system and the server.
Maintain & Upgrade	**Description**
WEB Configuration	Support of configurations through the WEB user interface.

Language	Chinese, English.
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB.
Tracking Test	Support of Ping and Tracert tests based on WEB.
SysLog Type	Three options available: ERROR, WARNING, INFO.

1.3 Hardware Description

The GSM gateway supports two LANs and adopts an external 12V power supply. See below for product appearance.

Figure 1-2 SMG4008 Front View

Figure 1-3 SMG4008 Rear View

The table below gives a detailed introduction to the interfaces, buttons and LEDs illustrated above:

Interface	Description
LAN	Amount: 2
	Type: RJ-45
	Bandwidth: 10/100 Mbps
	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
	Built-in Link indicator and ACTIVE indicator. For more details, refer to 1.4 Indicator

	Info
SIM Card Slot	Amount: 8
	Network Supported: GSM
Console Port	Amount: 1
	Type: RS-232
	Baud Rate: 115200bps
	Connector: RJ45 to DB-9 Connector
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
External Power Supply Interface	Provide the 12V voltage with positive inside and negative outside, and the current is larger than 3A
Button	**Description**
Reset Button	Restore the gateway to factory settings by pressing this button persistently for 3 seconds
LED	**Description**
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power cord well connected
Run Indicator	Indicates the running status. For more details, refer to 1.4 Indicator Info.
Alarm Indicator	Alarms the device malfunction. For more details, refer to 1.4 Indicator Info.
Link Indicator	The green LED on the right of LAN, indicating the network connection status.
ACT Indicator	The orange LED on the left of LAN, whose flashing tells the data are being transmitted.
Channel Indicator	1. When the channel is idle, the LED Lights up in green and keeps on; 2. When the channel is unavailable, the LED Lights up in red and keeps on; 3. When the channel is in use, the LED flashes.in green 4. When the channel module is disabled, the LED flashes in red.

For other hardware parameters, refer to Appendix A Technical Specifications.

1.4 Indicator Info

The GSM gateway is equipped with two indicators denoting the system's running status: Run Indicator (green LED) and Alarm Indicator (red LED). The table below explains the states and meanings of the two indicators.

LED	State	Description
Run Indicator	Go out	System is not yet started.
	Light up and flash fast	System is starting.
	Flash slowly	Device is normal.
Alarm Indicator	Go out	Device is normal.
	Light up	Upon startup: Device is normal. In runtime: Device is abnormal.
	Flash	Device is abnormal.

Note:

- The startup process consists of two stages: System Booting and Gateway Service Startup. The system booting costs about 1 minute and once it succeeds, both the run indicator and the alarm indicator light up. Then after the gateway service is successfully started and the device begins to work normally, the run indicator flashes and the alarm indicator goes out.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Go to Appendix C Technical/sales Support to find the contact way.

Chapter 2 Quick Guide

This chapter is intended to help you grasp the basic operations of the GSM gateway in the shortest time.

Step 1: Confirm that your packing box contains all the following things.

- GSM Gateway *1
- External 12V Power Adapter *1
- Network Cable *1
- Warranty Card *1
- Installation Manual *1

Step 2: Connect the network cable.

This product provides RJ-45 interfaces.

Step 3: Insert the SIM card.

The GSM gateway provides SIM card slot. You are required to insert the SIM card before using it.

Step 4: Power on and start the gateway.

To use the GSM gateway, you need an external power supply. Insert it to the power interface of the GSM gateway and power it on with 100~240V AC. See the figure below:

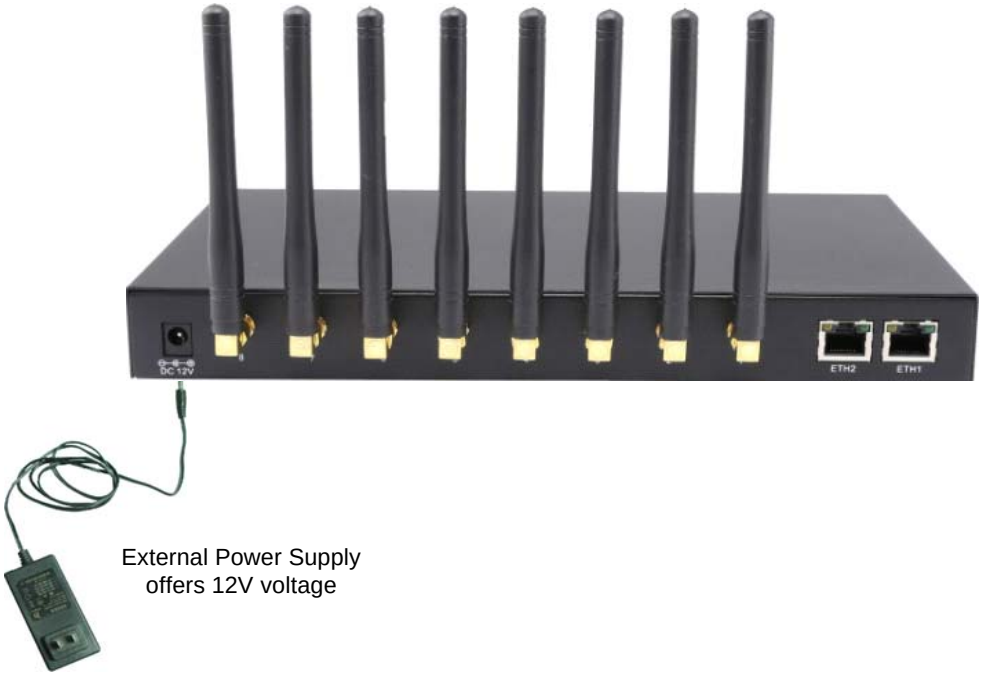

Figure 2-1 GSM Power Connection

Step 5: Log in the gateway.

Enter the original IP address (192.168.1.101) of the GSM gateway in the browser to go to the WEB interface of the gateway. The original username and password of the gateway are both 'admin'. For detailed instructions about login, refer to 3.1 System Login. We suggest you change the initial username and password via 'System Tools → Change Password' on the WEB interface as soon as possible after your first login. For detailed instructions about changing the password, refer to 3.10.4 Change Password. After changing the password, you are required to log in again.

Step 6: Modify IP address of the gateway.

You can modify the IP address of the gateway via 'Advanced Settings → Network' on the WEB interface to put it within your company's LAN. Refer to 3.5.1 Network for detailed instructions about IP modification. After changing the IP address, you shall log in the gateway again using your new IP address.

Step 7: Make phone calls.

Note: For your easy understanding and manipulation, all examples given in this step do not involve registration, that is, SIP initiates calls in a point-to-point mode.

Situation 1: Call from a station to an IP phone (Tel→IP)

1. Go to 'Advanced Settings → Dialing Rule' on the WEB interface and click the 'Add New' button to add a new dialing rule. Refer to 3.5.4 Dialing Rule for detailed instructions. Enter either a particular number or a string of 'x's to represent several random numbers. For example, 'xxx' denotes 3 random numbers. You may use the default value of 'Index' and are required not to leave 'Description' empty.

   Example: Set **Index** to **99**, fill in **Description** with **test** and configure **Dial Rule** to **123**.

2. Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add the corresponding ports to it. Refer to 3.7.2 Port Group for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

   Example: Provided the added port is Port1, check the checkbox before **Port1**, set **Index** to **1**, fill in **Description** with **test**, and keep the default values of other configuration items.

3. Go to 'Route Settings → Tel→IP' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to 3.8.3 Tel→IP for detailed instructions. Select the port group created in Step2 as 'Source Port Group' and fill in 'Destination IP' and 'Destination Port' with the IP address and the Port number you plan to call. You may use the default values of other configuration items and are required not to leave 'Description' empty.

   Example: Provided the remote IP address intended to call is 192.168.0.111 and the port is 5060. Set **Index** to **63**, **Source Port Group** to **1**, fill in **Description** with **test**, configure **Destination IP** to **192.168.0.111**, **Destination Port** to **5060**, and keep the default values of other configuration items.

4. Use an external phone to call the number of this SIM card, and then follow the cue tone to dial the number set in Step1 to ring the remote IP phone If you have set a particular number in Step 1, only this number you can dial; if you have set a string of 'x's, how many 'x's there are, how many random numbers you can dial.

   Example: The external phone dials the number of this SIM card, and then follows the cue tone to dial 123. Then the IP phone with the IP address 192.168.0.111 and the port 5060 will ring.

Situation 2: Call from an IP phone to a station (IP →Tel)

1. Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add the corresponding ports which are connected with stations to it. Refer to 3.7.2 Port Group for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

   Example: Provided the added port is Port1, check the checkbox before **Port1**, set **Index** to **1**, fill in **Description** with **test**, and keep the default values of other configuration items.

2. Go to 'Route Settings → IP→Tel' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to 3.8.2 IP→Tel for detailed instructions. Fill in 'Source IP' with the IP address which initiates the call and select the port group created in Step1 as 'Destination Port Group'. You may use the default values of other configuration items and required not to leave 'Description' empty.

   Example: Provided the IP address of the IP phone which initiates the call is 192.168.0.111.

Set **Index** to **63**, **Destination Port Group** to **1**, fill in **Description** with **test**, configure **Source IP** to **192.168.0.111**, and keep the default values of other configuration items.

3. Pick up the IP phone and call the IP address and port of the GSM gateway to make outgoing calls from the wireless channel.

   Example: Provided the IP address of the GSM gateway is 192.168.0.101, the port is 5060, use the IP phone to call the IP address 13529101232@192.168.0.101 and then the first idle wireless channel in the port group of step 2 will make an outgoing call to 13529101232.

Special Instructions:

- As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes are never jammed.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Otherwise it may lead to a drop in performance or unexpected errors.

Chapter 3 WEB Configuration

3.1 System Login

Type the IP address into the browser and enter the login interface. See Figure 3-1.

Windows Security
The server 192. 168. 1. 101 at SMG requires a username and password.
Warning: This server is requesting that your username and password be sent in an insecure manner (basic authentication without a secure connection).
User name
Password
Remember my credentials
OK
Cancel

Figure 3-1 Login Interface

The gateway only serves one user, whose original username and password are both 'admin'. You can change the username and the password via 'System Tools → Change Password' on the WEB interface. For detailed instructions, refer to 3.10.4 Change Password.

After login, you can see the main interface as below.

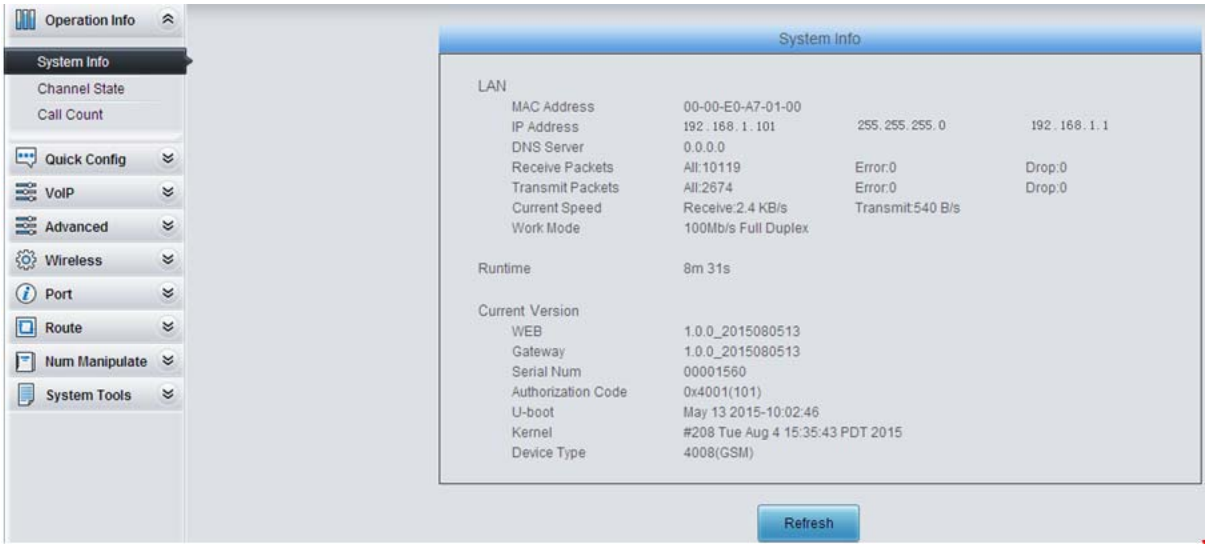

Figure 3-2 Main Interface

3.2 Operation Info

Operation Info includes three parts: ***System Info***, ***Channel State*** and ***Call Count***, showing the current running status of the gateway. See Figure 3-3.

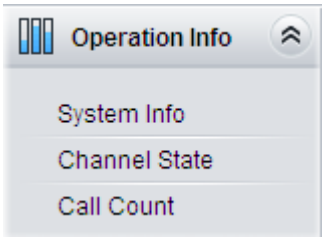

Figure 3-3 Operation Info

3.2.1 System Info

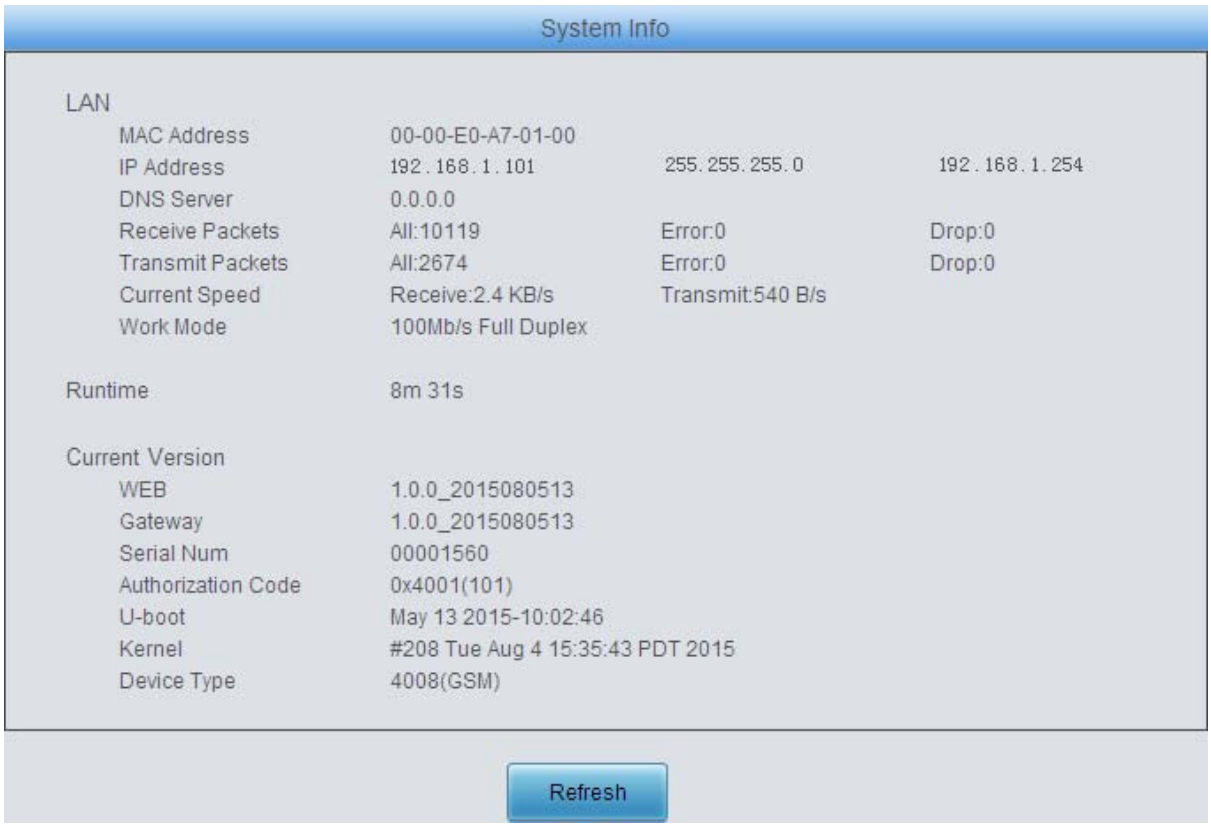

Figure 3-4 System Info Interface

See Figure 3-4 for the system info interface. You can click ***Refresh*** to obtain the latest system information. The table below explains the items shown in Figure 3-4.

Item	Description
MAC Address	MAC address of LAN.
IP Address	The three parameters from left to right are IP address, subnet mask and default gateway of LAN.
DNS Server	DNS server address of LAN.
Receive Packets	The amount of receive packets after the gateway's startup, including three options: All, Error and Drop.

Transmit Packets	The amount of transmit packets after the gateway's startup, including three options: All, Error and Drop.
Current Speed	Show the current speed of data receiving and transmitting.
Work Mode	Show the work mode of the network, including four modes: 10 Mbps Half Duplex, 10 Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex.
Runtime	Time of the gateway keeping running normally after startup, which will be automatically updated.
WEB	Current version of the WEB interface.
Gateway	Current version of the gateway service.
Serial Num	Unique serial number of a GSM wireless gateway.
Authorization Code	The authorization codes vary from different GSM modules.
U-boot	Current version of Uboot.
Kernel	Current version of the system kernel on the gateway.
Device Type	Type of the wireless gateway.

3.2.2 Channel State

Channel State
Channel Type IMSI IMEI State Direction CallerID CalleeID Signal GSM Connection SIP Reg Status
1 GSM --- 866699020017804 Unusable --- --- --- Disconnect Unregistered
2 GSM 460029900309534 866699020017788 Idle --- --- --- Connect Unregistered
3 GSM --- 866699020017945 Unusable --- --- --- Disconnect Unregistered
4 GSM --- 866699020017648 Unusable --- --- --- Disconnect Unregistered
5 GSM --- 866699020018265 Unusable --- --- --- Disconnect Unregistered
6 GSM --- 866699020018257 Unusable --- --- --- Disconnect Unregistered
7 GSM --- --- Unusable --- --- --- Disconnect Unregistered
8 GSM --- --- Unusable --- --- --- Disconnect Unregistered

Figure 3-5 Channel State Interface

See Figure 3-5 for the channel state interface where shows the channel type, the channel state for each channel on the gateway. The table below explains the items shown in Figure 3-5.

Item	Description
Channel	Channel number on the device.
Type	Channel type on the device. So far, only GSM type is supported.
IMSI	International Mobile Subscriber Identification Number, the unique identity of the SIM card.
IMEI	International Mobile Equipment Identity.
State	Displays the channel state in real time. You can move the mouse onto the channel state icon for detailed state information.

State	Icon	Description
Idle		The channel is available.
Off-hook		The channel picks up the call.
Wait Answer		The channel receives the ringback tone and is waiting for the called party to pick up the phone.
Ringing		The channel is in the ringing state.
Talking		The channel is in a conversation.

	Dialing		The channel is dialing.
	Pending		The channel is in the pending state.
	Internal State		Internal state of the channel.
	Unusable		The channel is unavailable.
Direction	Displays the direction of the call on channel.		
CallerID	Displays the CallerID of the call on channel.		
CalleeID	Displays the CalleeID of the call on channel.		
Signal	Displays the signal intensity of the GSM module.		
GSM Connection	Displays the connection status of the GSM module.		
SIP Reg Status	Displays the registration status of the port.		

3.2.3 Call Count

Figure 3-6 Call Count Interface

See Figure 3-6 for the call count Interface. The above list shows the detailed information about all the calls counted from the startup of the gateway service to the latest open or refresh of this interface. You can click ***Refresh*** to obtain the current call count information. The table below explains the items shown in Figure 3-6.

Item	Description
Call Direction	A condition for call count, two options available: *IP→Tel* and *Tel→IP*.
Total Calls	Total number of calls in a specified call direction.
Successful Calls	Total number of successful calls in conversation.
Busy	Total number of calls which fail as the called party has been occupied and replies a busy message.
No Answer	Total number of calls which fail as the called party does not pick up the call in a long time or the calling party hangs up the call before the called party picks it up.
Call Forward	Total number of calls which have been forwarded.
Routing Failure	Total number of calls which fail because no routing rules are matched.
Dialing Failure	Total number of calls which fail as the called party number does not conform to the dialing rule or due to dialing timeout.
Unknown Failure	Total number of calls which fail due to unknown reasons.

3.3 Quick Config

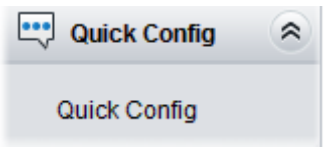

Figure 3-7 Quick Config Interface

See Figure 3-7 for the Quick Config interface. Follow the gateway Quick Configuration wizard and you can easily complete the settings on network, SIP and Port. The gateway can work normally after configuration.

See Figure 3-8 for the Quick Config-Network Settings interface. Refer to 3.5.1 Network for detailed settings. After configuration, click ***Next*** to enter the SIP Settings interface.

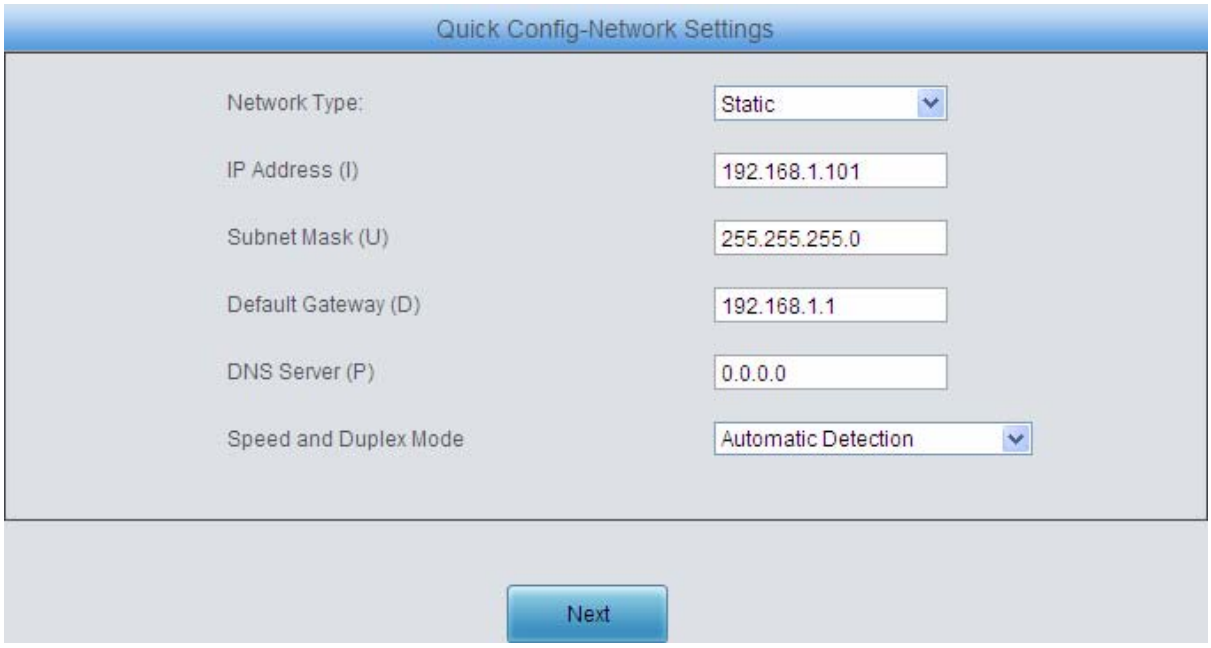

Figure 3-8 Quick Config-Network Settings Interface

See Figure 3-9 for the Quick Config-SIP Settings interface. The configuration items on this interface are the same as those on the SIP interface. Refer to 3.4.1 SIP for detailed settings. You are required to fill with the information about the registrar if the gateway must be registered. After configuration, click ***Back*** to go back to the Network Settings interface; click ***Next*** to enter the Port Settings interface.

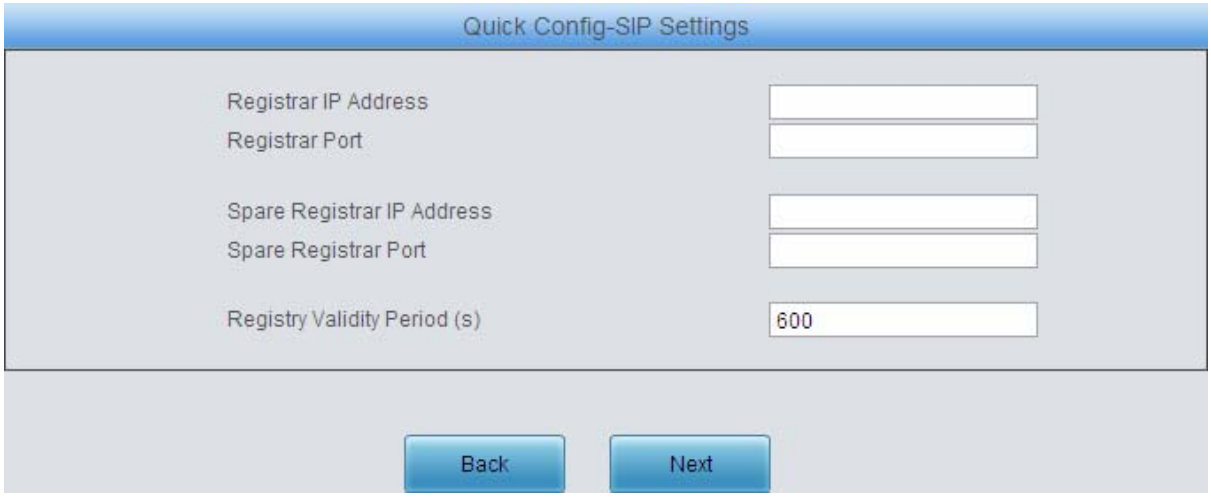

Figure 3-9 Quick Config-SIP Settings Interface

See Figure 3-10 for the Port Settings interface. The configuration items on this interface are the same as those on the Port interface. Refer to 3.7.1 Port for detailed settings. After configuration, click ***Back*** to go back to the SIP Settings interface; click ***Next*** to enter the Quick Config-Completion interface.

Port Settings — Batch Modify

Port	Type	SIP Account	Authentication Username	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Modify
1	GSM	8001	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	
2	GSM	8002	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	
3	GSM	8003	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	
4	GSM	8004	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	
5	GSM	8005	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	
6	GSM	8006	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	
7	GSM	8007	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	
8	GSM	8008	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	

Back Next

Figure 3-10 Port Settings Interface

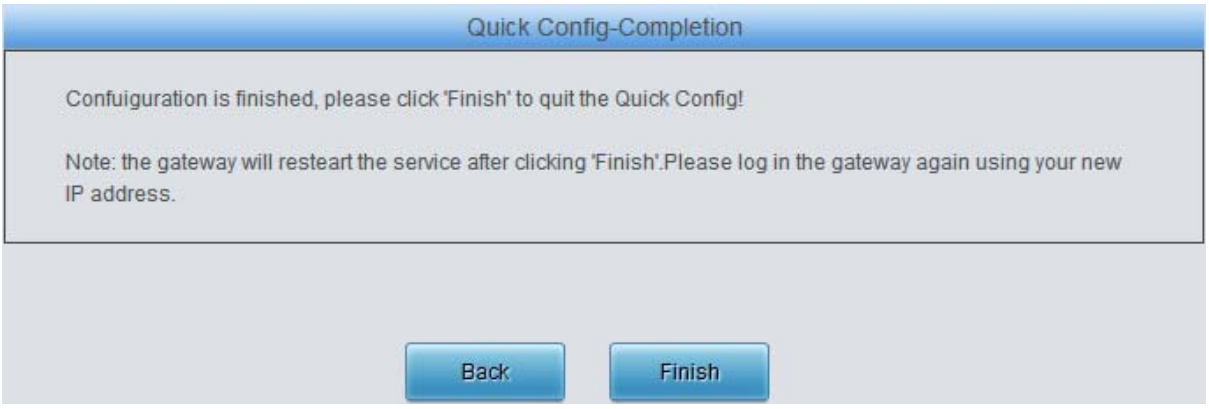
Quick Config-Completion
Confuiguration is finished, please click 'Finish' to quit the Quick Config!
Note: the gateway will resteart the service after clicking 'Finish'.Please log in the gateway again using your new IP address.
Back
Finish

Figure 3-11 Quick Config-Completion Interface

Click ***Back*** to go back to the Port Settings interface; click ***Finish*** to finish the Quick Config wizard and now the gateway can work normally with basic configuration.

3.4 VoIP Settings

VoIP Settings includes six parts: ***SIP***, ***SIP Compatibility***, ***SIP Station***, **SIP Server**, ***NAT Setting*** and ***Media***. See Figure 3-12. ***SIP Settings*** is used to configure the general SIP parameters, ***SIP Compatibility*** is used to set which SIP servers and SIP messages will the gateway be compatible with, ***SIP Station*** is to set the basic information of the SIP station, ***SIP Server*** is to set the basic information of the SIP server, ***NAT Setting*** is used to configure the parameters for NAT, and ***Media Settings*** is to set the RTP port and the payload type.

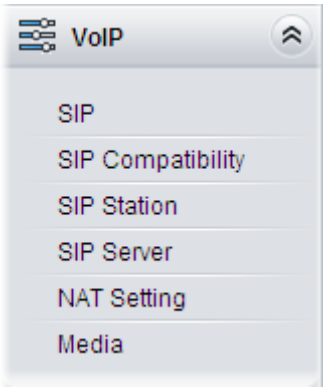

Figure 3-12 VoIP Settings

3.4.1 SIP

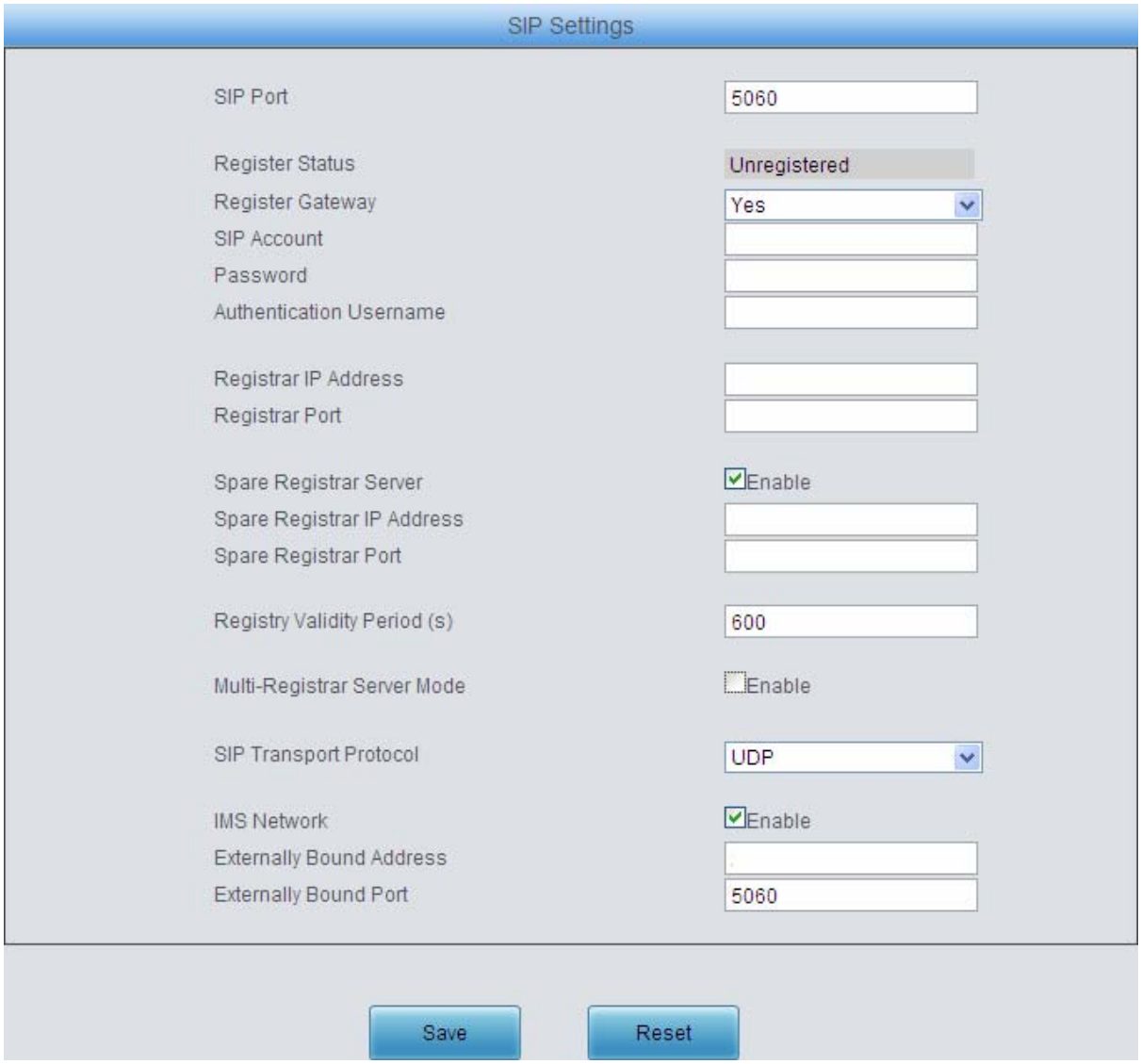

Figure 3-13 SIP Settings Interface

See Figure 3-13 for the SIP settings interface where you can configure the general SIP parameters. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to 3.10.7 Restart for detailed instructions. The table below explains the items shown in Figure 3-13.

Item	Description
SIP Port	Monitoring port of SIP signaling. The value range of it must be grater than 1024 and less than 65535, with the default value of 5060.
Register Status	Registration status of the gateway. When ***Register Gateway*** is set to *No*, the value of this item is *Unregistered*; when ***Register Gateway*** is set to *Yes*, the value of this item is either *Failed* or *Registered*.
Register Gateway	Sets whether to register the gateway as a whole. The default value is *No*. Only when this configuration is set to *Yes* can you see the configuration items ***SIP Account*** and ***Password***.

SIP Account	When the gateway initiates a call to SIP, this item corresponds to the username of SIP.
Password	Registration password of the gateway. To register the gateway to SIP, both configuration items ***SIP Account*** and ***Password*** should be filled in.
Authentication Username	Authentication username for registration.
Registrar IP Address	Address of the registry server for the gateway to register.
Registrar Port	Signaling port of the registry server.
Spare Registrar Server	Check the enable checkbox to enable the spare registrar server. By default, it is ***disabled***.
Spare Registrar IP Address	Address of the spare registry server for the gateway to register. The gateway will enable the spare registrar server if the master registrar server has no reply, or the master server is detected with no response in case the item ***Detection Server Cycle*** is enabled.
Spare Registrar Port	Signaling port of the spare registry server.
Registry Validity Period	Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. This configuration item is valid only when ***Register Gateway*** is set to *Yes*. Range of value: 10~3600, calculated by s, with the default value of 600.
Multi-Registrar Server Mode	Tick the checkbox before to enable the multi-registrar server mode. By default, it is *disabled*.
SIP Transport Protocol	There are two modes *UDP* and *TCP* available for running the SIP protocol. The default value is *UDP*.
IMS Network	Once this feature is enabled, the gateway will send signaling messages to the corresponding externally bound address and port when it registers to the server. By default, this feature is *disabled*. Only when this feature is *enabled* will these items ***Externally Bound Address***, ***Externally Bound Port*** and ***Authentication Username*** be shown.
Externally Bound Address	Externally bound IP address for registration.
Externally Bound Port	Externally bound port for registration.

3.4.2 SIP Compatibility

See Figure 3-14 for the SIP Compatibility interface where you can configure the SIP parameters to determine which SIP servers and SIP messages will the gateway be compatible with. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

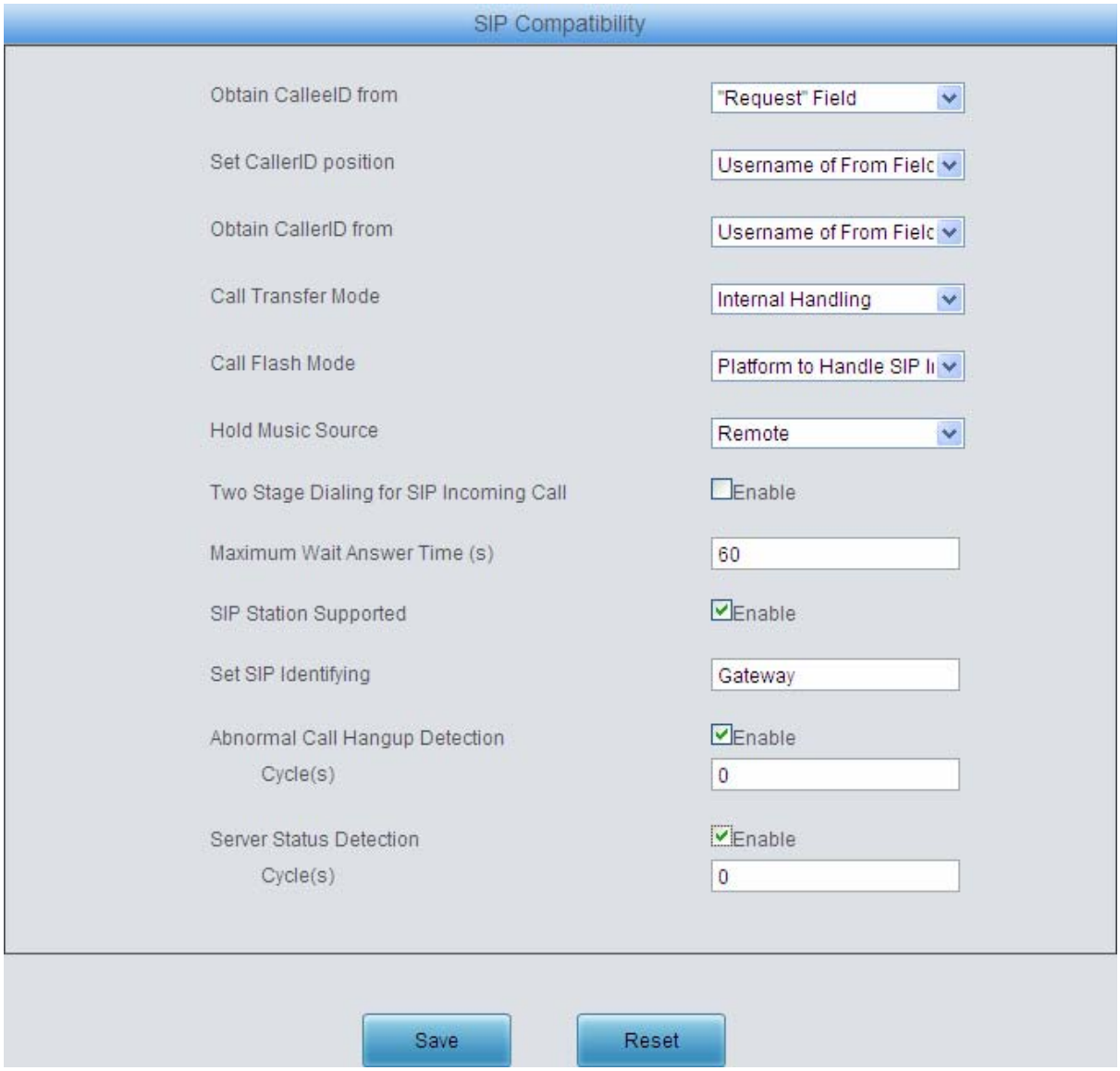

Figure 3-14 SIP Compatibility Setting Interface

The table below explains the items shown in Figure 3-14.

Item	Description
Obtain CalleeID from	There are two optional ways to obtain the called party number: from "*To" Field* and from "*Request" Field*. The default value is "*Request" Field*.
Set CallerID Position	There are two options to set the position of the calling party number: "*Displayname of From Field*" and "*Username of From Field*". The default value is "*Username of From Field*".
Obtain CallerID from	There are two optional ways to obtain the calling party number: from "*Displayname of From Field*" and from "*Username of From Field*". The default value is "*Username of From Field*".
Call Transfer Mode	There are two optional ways to deal with call transfer: *Internal Handling* and *Platform to Handle SIP Info*. The default value is *Internal Handling*.
Call Flash Mode	There are two optional ways to deal with call flash: *Internal Handling* and *Platform to Handle SIP Info*. The default value is *Internal Handling*.

Hold Music Source	Sets the source of the hold music, with the default value of ***Remote***, This feature gets valid only when you choose the mode *Platform to Handle SIP Info*.
Two Stage Dialing for SIP Incoming Call	Once this feature is enabled, the incoming call from SIP should perform the two stage dialing operation. By default this feature is disabled.
Maximum Wait Answer Time	Sets the maximum time for the SIP channel to wait for the answer from the called party of the outgoing call it initiates. If the call is not answered within the specified time period, it will be canceled by the channel automatically. The default value is 60, calculated by s.
SIP Station Supported	Once this feature is enabled, a SIP terminal can be registered to the gateway to become a SIP station. By default this feature is disabled.
Set SIP Identifying	Sets the SIP identifying content in the SIP call message. The default setting is *Gateway*.
Abnormal Call Hangup Detection	Sets the interval between checks of the remote end's abnormal hangup, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 10s if this feature is necessary to be used.
Server Status Detection	The interval of sending a heartbeat packet to detect the master registrar server status, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 15s if this feature is necessary to be used.

3.4.3 SIP Station

A SIP terminal can be registered to the gateway to become a SIP station. Tick the option of '***SIP Station Supported***' on 3.4.2 SIP Compatibility interface, and you will see the item SIP Station on the VoIP Settings menu. Click '***SIP Station***' to go into the SIP Station interface. By default, there is no available SIP station. See Figure 3-15 below.

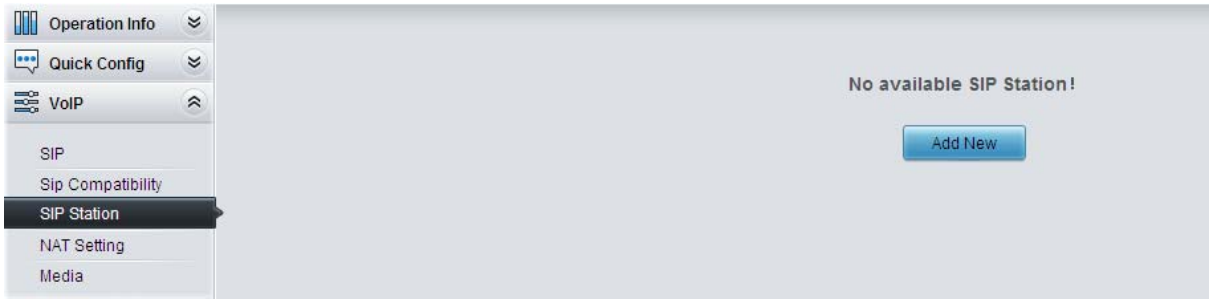

Figure 3-15 SIP Station Setting Interface

Click ***Add New*** to add SIP stations manually. See Figure 3-16. You can configure basic SIP station information on this interface. The bound port to a SIP station must be a wireless port and unique. The username must be the same as that used to register the SIP terminal to the gateway.

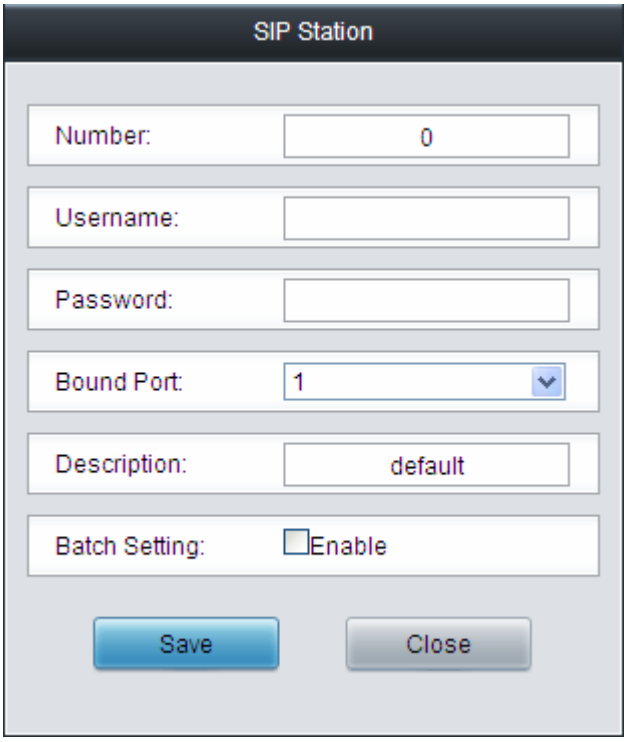

Figure 3-16 Add New SIP Station

The table below explains the items shown above:

Item	Description
Number	The logical number for a SIP station to register to the gateway.
Username	The username used to register a SIP station to the gateway.
Password	The password used to register a SIP station to the gateway.
Bound Port	The wireless port which is bound to the SIP station.
Description	It is user-defined, with the default value of *default*.
Batch Setting	Used to set multiple SIP stations at the same time.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Close*** to cancel the settings. See Figure 3-17 for the applied SIP station information.

Figure 3-17 SIP Station Interface

Click ***Modify*** in the above figure to modify the configuration of the SIP station. See Figure 3-18. The configuration items on this interface are the same as those on the ***Add New SIP Station*** interface.

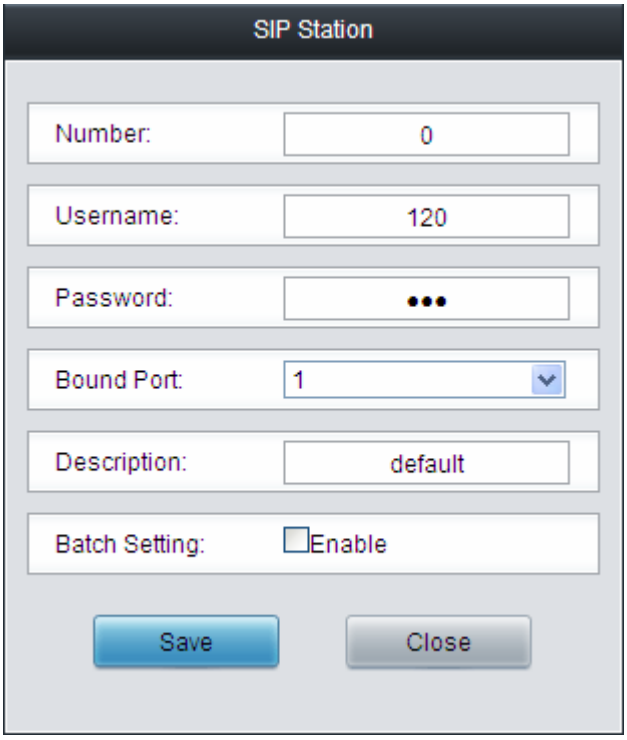

Figure 3-18 SIP Station Modification Interface

To delete a SIP station, check the checkbox before the corresponding index in Figure 3-17 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all SIP stations at a time, click the ***Clear All*** button in Figure 3-17.

3.4.4 SIP Server

The gateway supports the multi-registrar server feature. Enable the feature of '***Multi-Registrar Server Mode***' on the SIP interface (see 3.4.1 SIP) and you will see the item SIP Server under the VoIP Settings menu. Click '***SIP Server***' to go into the SIP Server interface. By default, there is no available SIP server. See Figure 3-19 below.

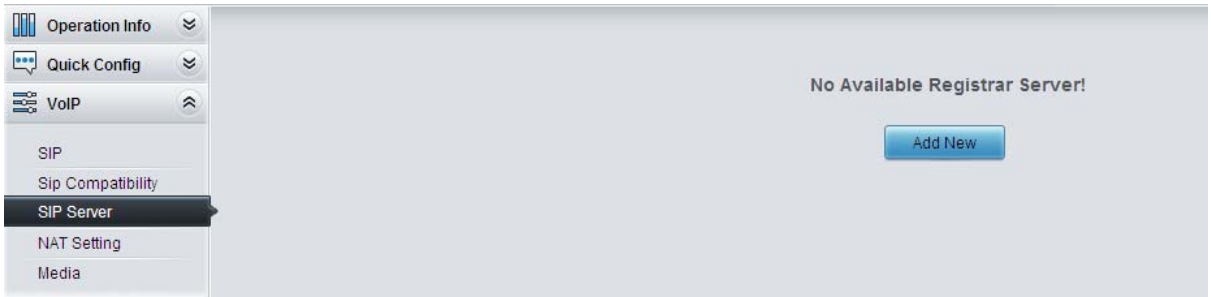

Figure 3-19 SIP Server Interface

Click ***Add New*** to add SIP servers manually. See Figure 3-20. You can configure basic SIP server information on this interface.

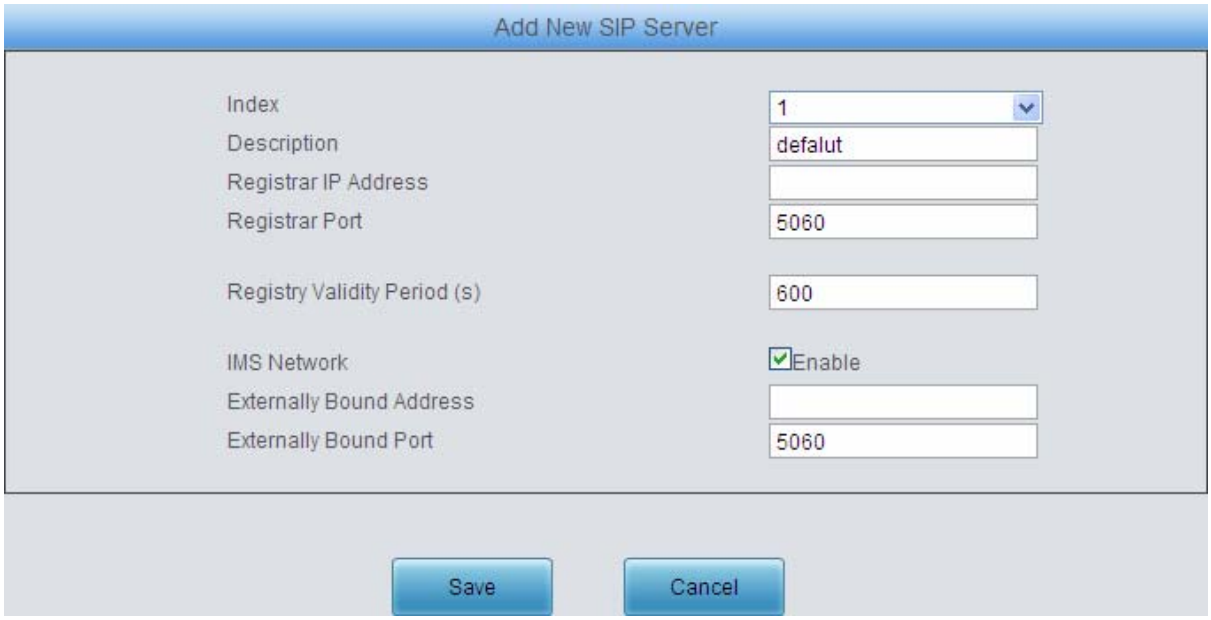

Figure 3-20 Add New SIP Server

All the items except Index and Description are the same as those on the SIP interface (3.4.1 SIP).

Item	Description
Index	The index of each SIP server. The gateway supports up to 8 SIP servers.
Description	More information about each SIP server, with the default value of *default*.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Cancel*** to cancel the settings. See Figure 3-21 for the SIP server management interface.

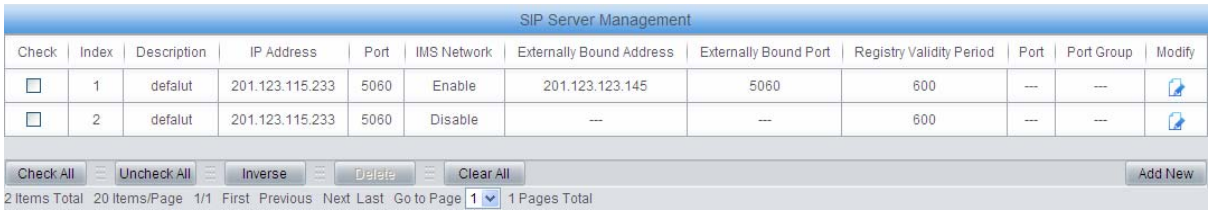

Figure 3-21 SIP Server Management

Click ***Modify*** in the above figure to modify the configuration of the SIP server. See Figure 3-22.

The configuration items on this interface are the same as those on the ***Add New SIP Server*** interface.

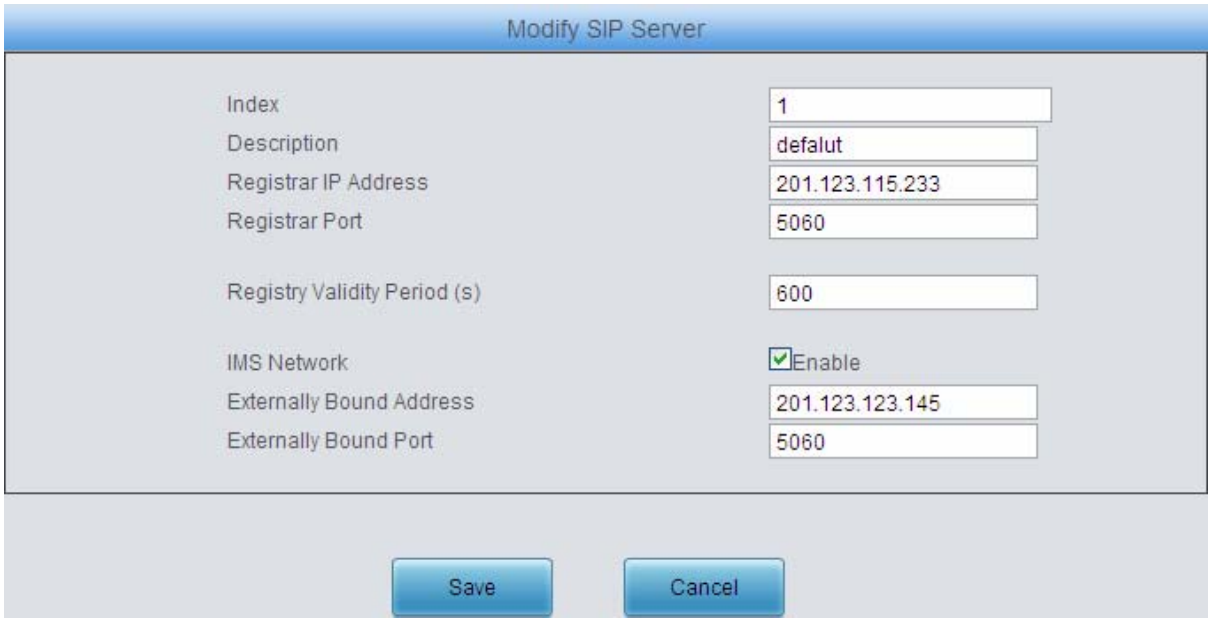

Figure 3-22 SIP Server Modification Interface

To delete a SIP server, check the checkbox before the corresponding index in Figure 3-21 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all SIP servers at a time, click the ***Clear All*** button in Figure 3-21.

3.4.5 NAT Setting

See Figure 3-23 for the NAT setting interface where you can configure the parameters for NAT. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

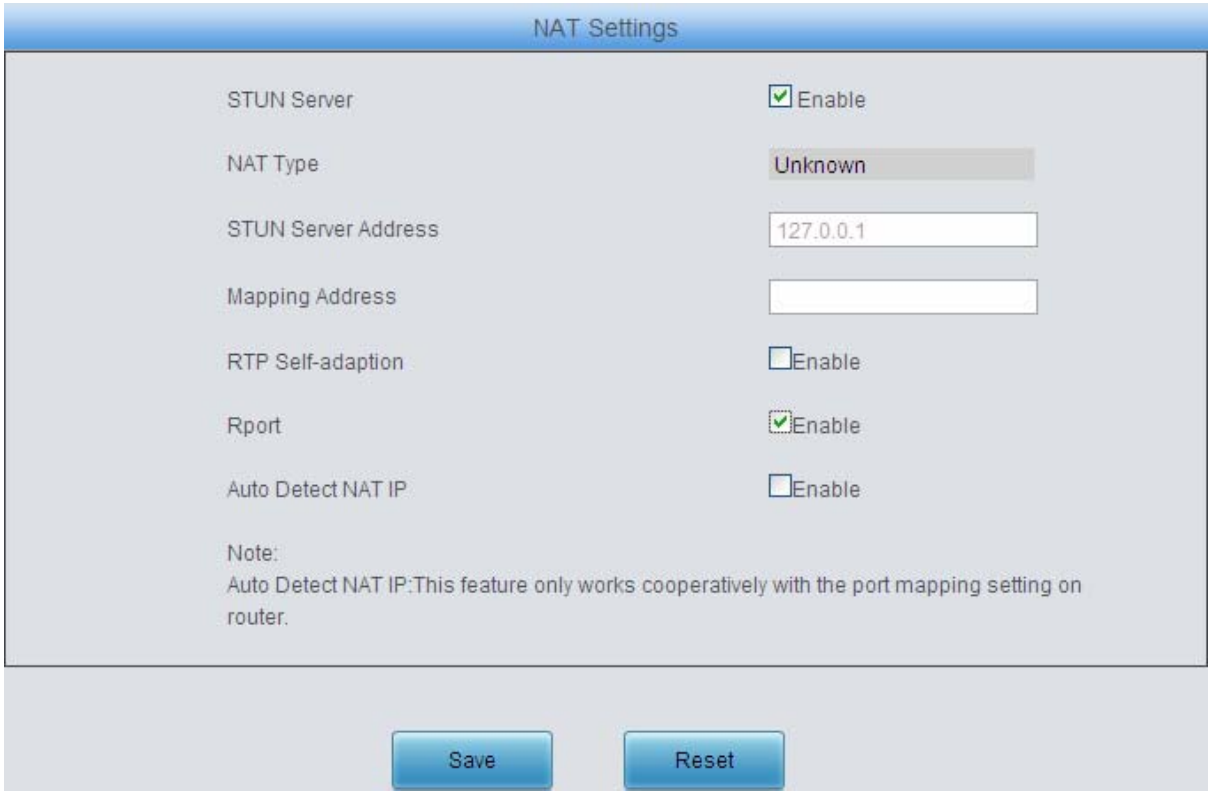

Figure 3-23 NAT Setting Interface

The table below explains the items shown in Figure 3-23.

Item	Description
STUN Server	Sets whether to enable the STUN server for NAT traversal. By default the STUN server is disabled.
NAT Type	Detected NAT (Network Address Translation) type. The gateway will return the NAT type automatically in case ***STUN Server*** is enabled. It includes 9 types: unknown; no NAT; ConeNat; RestrictedNat; PortRestrictedNat; Symmetric NAT; Symmetric NAT with firewall; can't detect over (fail to send detect message) and fail to detect (No reply from the stun server).
STUN Server Address	Address of the server for STUN traversal.
Mapping Address	It should be filled in when there exists NAT or other mapping relationships which leads to the failure of direct communication between the gateway and the destination address, so as to ask the remote end to send signaling messages or voice data to it during the signaling or voice communication between the gateway and the destination. **Note:** Once this item is filled out, it will be used as the first choice even if Rport and NAT IP are enabled.
RTP Self-adaption	When this feature is enabled, the RTP reception address or port carried by the signaling message from the remote end, if not consistent with the actual state, will be updated to the actual RTP reception address or port. By default, this feature is *disabled*.
Rport	When this feature is enabled, a corresponding Rport field will be added to the Via message of SIP. The default value is *enabled*.
Auto Detect NAT IP	When this feature is enabled, the gateway will parse the corresponding address and port in the message returned by Rport so as to use them for the following communication. By default, this feature is *disabled*. **Note:** This feature gets valid only when Rport is enabled.

3.4.6 Media

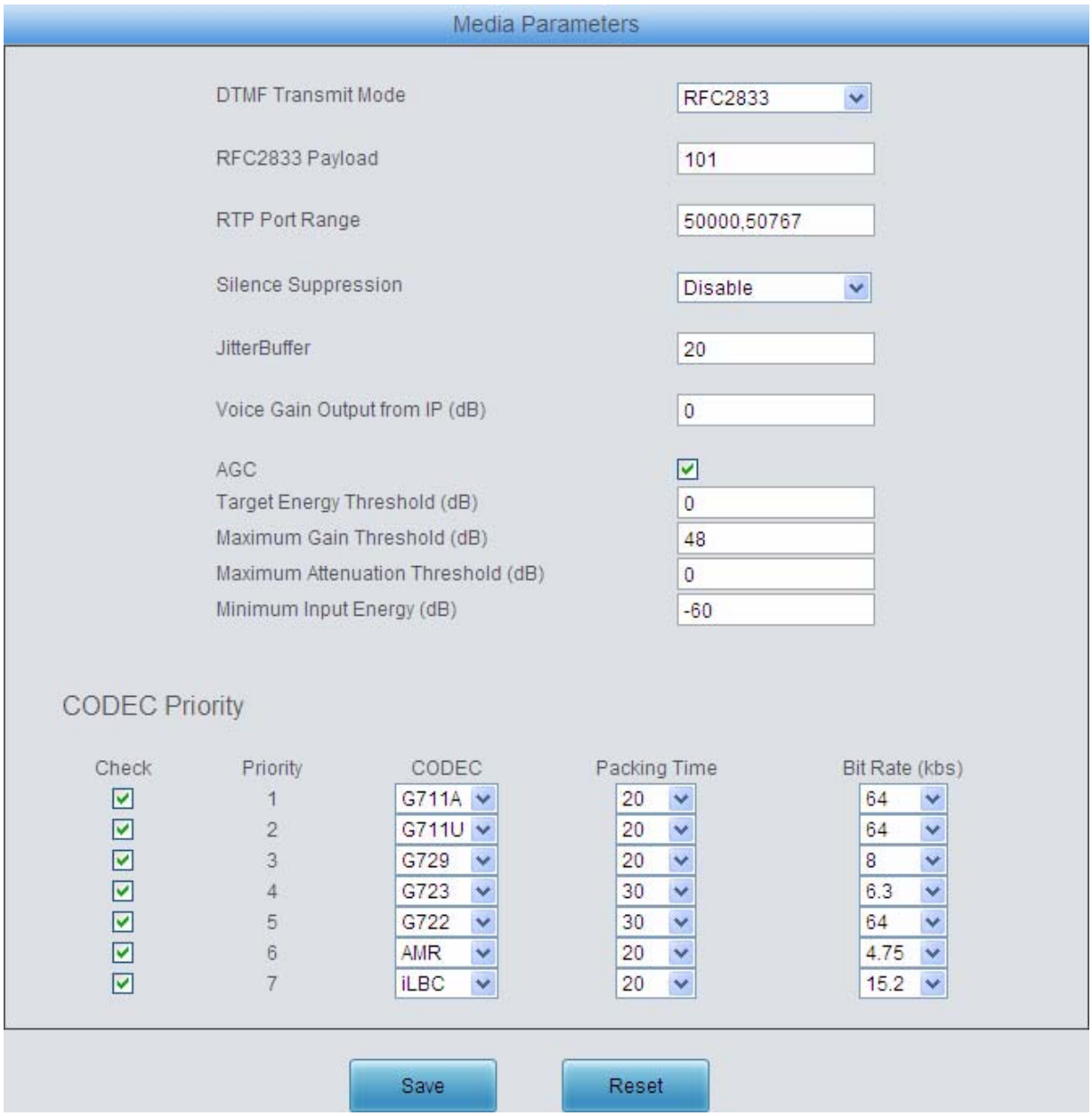

Figure 3-24 Media Settings Interface

See Figure 3-24 for the media settings interface where you can configure the RTP port and payload type depending on your requirements. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to 3.10.7 Restart for detailed instructions. The table below explains the items shown in Figure 3-24.

Item	Description
DTMF Transmit Mode	Sets the transmit mode for the IP channel to send DTMF signals. The optional values are *RFC2833*, *In-band* and *Signaling*, with the default value of *RFC2833*.
RFC2833 Payload	Payload of the RFC2833 formatted DTMF signals on the IP channel. Range of value: 90~127, with the default value of 101.

RTP Port Range	Supported RTP port range for the IP end to establish a call conversation, with the lower limit of 10000 and the upper limit of 60000 and the difference between larger than 480. The default value is 50000-50767.
Silence Suppression	Sets whether to send comfort noise packets to replace RTP packets or never to send RTP packets to reduce the bandwidth usage when there is no voice signal throughout an IP conversation. The optional values are *Enable* and *Disable*, with the default value of *Disable*.
JitterBuffer	Acceptable jitter for data packets transmission over IP, which indicates the buffering capacity. A larger JitterBuffer means a higher jitter processing capability but as well as an increased voice delay, while a smaller JitterBuffer means a lower jitter processing capability but as well as a decreased voice delay. Range of value: 20~200, calculated by ms, with the default value of 20.
Voice Gain Output from IP	Adjusts the gain of the voice output from IP. Range of value: -24~12, calculated by dB, with the default value of 0.
AGC	If the AGC (Automatic Gain Control) feature is enabled, the gateway will automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals.
Target Energy Threshold	Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0.
Maximum Gain Threshold	Set the maximum gain threshold that will be applied to the signal. Range of value: 0~48, calculated by dB, with the default value of 48.
Maximum Attenuation Threshold	Set the maximum attenuation that will be applied to the signal. Range of value: -42~0, calculated by dB, with the default value of 0.
Minimum Input Energy	Set the minimum threshold for the energy processed by AGC. Signals below this threshold will not be processed by AGC. Range of value: -60~ -25, calculated by dB, with the default value of -60.

<table>
<tr><td>CODEC Priority</td><td>
Supported CODECs and their corresponding priority for the IP end to establish a call conversation. The table below explains the sub-items:
<table>
<tr><th>Sub-item</th><th>Description</th></tr>
<tr><td>Priority</td><td>Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.</td></tr>
<tr><td>CODEC</td><td>Three optional CODECs are supported: G711A, G711U, G729A/B, G723, G722, AMR and iLBC.</td></tr>
<tr><td>Packing Time</td><td>Time interval for packing an RTP packet, calculated by ms.</td></tr>
<tr><td>Bit Rate</td><td>The number of thousand bits (excluding the packet header) that are conveyed per second.</td></tr>
</table>
By default, all of the seven CODECs are supported and ordered G711A, G711U, G729A/B, G723, G722, AMR and iLBC by priority from high to low.
The packing time and bit rate supported by different CODECs are listed in the table below. Those values in bold face are the default values.
<table>
<tr><th>COEDC</th><th>Packing Time (ms)</th><th>Bit Rate (kbps)</th></tr>
<tr><td>G711A</td><td>10 / 20 / 30 / 40 / 60</td><td>64</td></tr>
<tr><td>G711U</td><td>10 / 20 / 30 / 40 / 60</td><td>64</td></tr>
<tr><td>G729A/B</td><td>10 / 20 / 30 / 40 / 60</td><td>8</td></tr>
<tr><td>G723</td><td>30 / 60</td><td>5.3 / 6.3</td></tr>
<tr><td>G722</td><td>10 / 20 / 30 / 40</td><td>64</td></tr>
<tr><td>AMR</td><td>20 / 40 / 60</td><td>4.75</td></tr>
<tr><td rowspan="2">iLBC</td><td>20 / 40</td><td>15.2</td></tr>
<tr><td>30 / 60</td><td>13.3</td></tr>
</table>
</td></tr>
</table>

3.5 Advanced Settings

Advanced Settings includes eight parts: ***Network***, ***System Param***, ***Service Config***, ***Dialing Rule***, ***Function Key***, ***Cue Tone***, ***QoS*** and ***Tone Generator***. See Figure 3-25. ***Network*** is used to configure the general properties of the network port; ***System Param*** is used to configure some properties of the system; ***Service Config*** is used to configure some properties which corresponds to the service; ***Dialing Rule*** is used to set the judging conditions for dialing; ***Function Key*** is used to set a cluster of combination keys for you to query a related number; ***Cue Tone*** is used to set the gateway language for playing voice and the voice file used for the two-stage dialing; ***QoS*** uses the differentiated services technology to increase the gateway's service quality; ***Tone Generator*** is used to configure some properties of tones sent from gateway.

Figure 3-25 Advanced Settings

3.5.1 Network

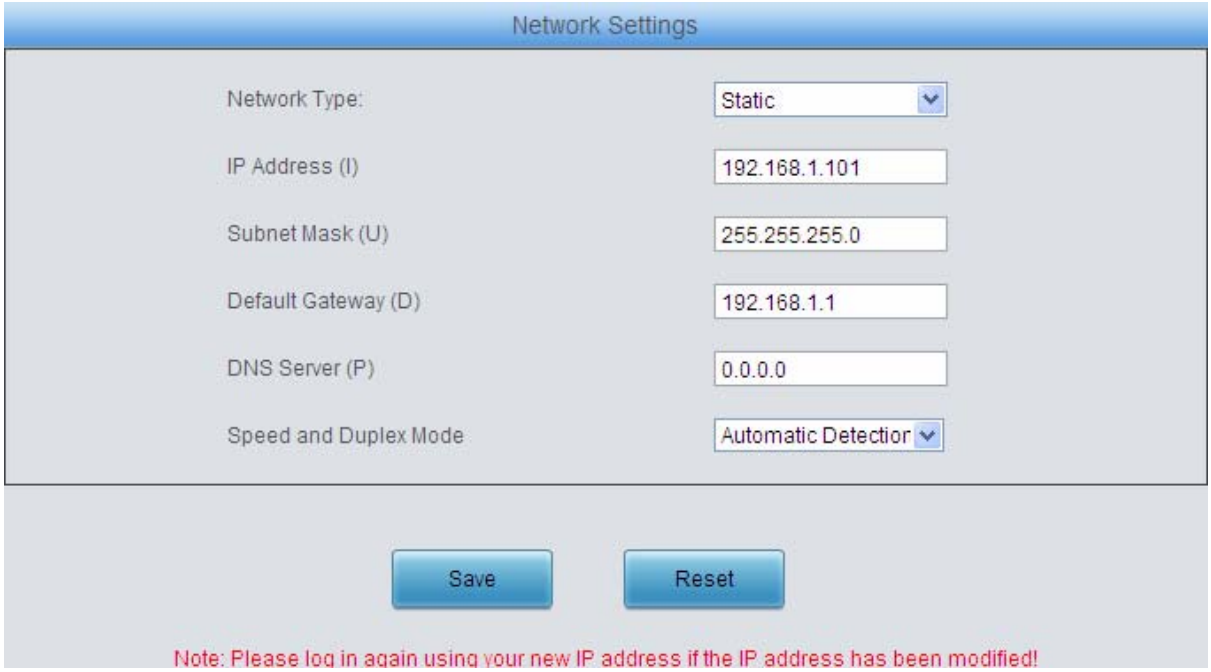

Figure 3-26 Network Settings Interface

See Figure 3-26 for the network settings interface. A gateway has two LANs which can be configured with the same network type, IP address, subnet mask, default gateway and DNS server to realize the feature of hot backup. There are three options in type: Static, DHCP and PPPoE.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations. After changing the IP address, you shall log in the gateway again using your new IP address.

3.5.2 System Param

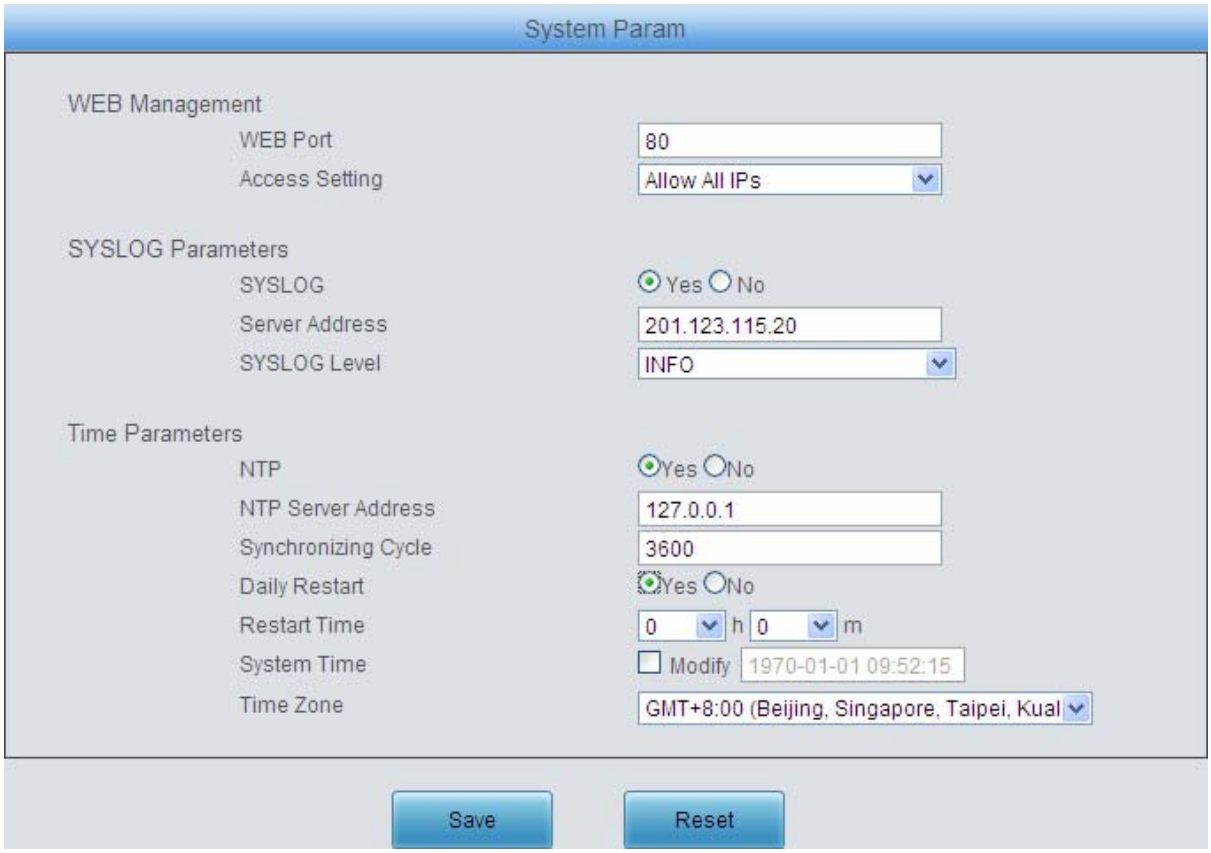

Figure 3-27 System Parameters Setting Interface

See Figure 3-27 for the System Parameters Setting interface. The table below explains the items shown in the above figure.

Item	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
Access Setting	Sets the IP addresses which can access the gateway via WEB. By default, all IPs are allowed. You can set an IP whitelist to allow all IPs within it to access the gateway freely. Also you can set an IP blacklist to forbid all IPs within it to access the gateway.
SYSLOG	Sets whether to enable SYSLOG. It is required to fill in ***SYSLOG Server Address*** and ***SYSLOG Level*** in case SYSLOG is enabled. By default, ***SYSLOG*** is disabled.
Server Address	Sets the SYSLOG server address for log reception.
SYSLOG Level	Sets the SYSLOG level. There are three options: *ERROR*, *WARNING*, *INFO* and *DEBUG*. The default value is *INFO*.
NTP	Sets whether to enable the NTP time synchronization feature. It is required to fill in ***NTP Server Address***, ***Synchronizing Cycle*** and ***Time Zone*** in case NTP is enabled. By default, ***NTP*** is disabled.
NTP Server Address	Sets the Server address for NTP time synchronization.
Synchronizing Cycle	Sets the cycle for NTP time synchronization. The default value is 3600.
Daily Restart	Sets whether to restart the gateway regularly every day at the preset ***Restart Time***. By default, this feature is disabled.

Restart Time	Sets the time to restart the gateway regularly.
System Time	The system time. Check the checkbox before ***Modify*** and change the time in the edit box.
Time Zone	The time zone of the gateway.

3.5.3 Service Config

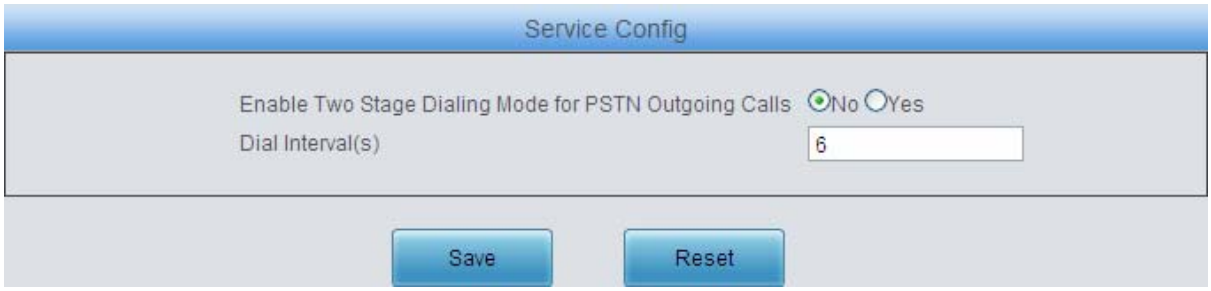

Figure 3-28 Service Config Interface

See Figure 3-28 for the Service Config interface. The table below explains the items shown in the above figure.

Item	Description
Enable Two Stage Dialing Mode for PSTN Outgoing Calls	Sets whether to enable the two stage dialing mode for PSTN outgoing calls. Under this mode, an outgoing call from a wireless port will go into the IVR system. Then IVR will play a speech prompt "Please dial the number". If you fail to input the number before IVR finishes the third repeat of the prompt, the wireless channel will hang up the call automatically; otherwise, it will make an outgoing call to the number.
Dial Interval	Sets the largest interval between two digits of a dialing number. Range of value: 1~10, calculated by s, with the default value of 6. In case your dialing rules do not include "**.**", the call will fail if there is no digit dialed or no dialing rule matched during this interval; in case your dialing rules include "**.**", the gateway will wait until this interval ends and match to the dialing rule "**.**" if there is no digit dialed or no other dialing rule matched during this interval.

3.5.4 Dialing Rule

Considering efficiency, it is not acceptable that the gateway reports to the PBX or relevant devices every time it receives a number. Instead, we hope that the gateway can automatically judge the received number to see if it meets the set rule, if it is complete and if it is qualified to make outgoing calls. Therefore, a whole dialing plan, which consists of multiple dialing rules specifying the auto judging conditions, is required. Each dialing rule has a priority, which is used to restrict the sequence and avoid conflict.

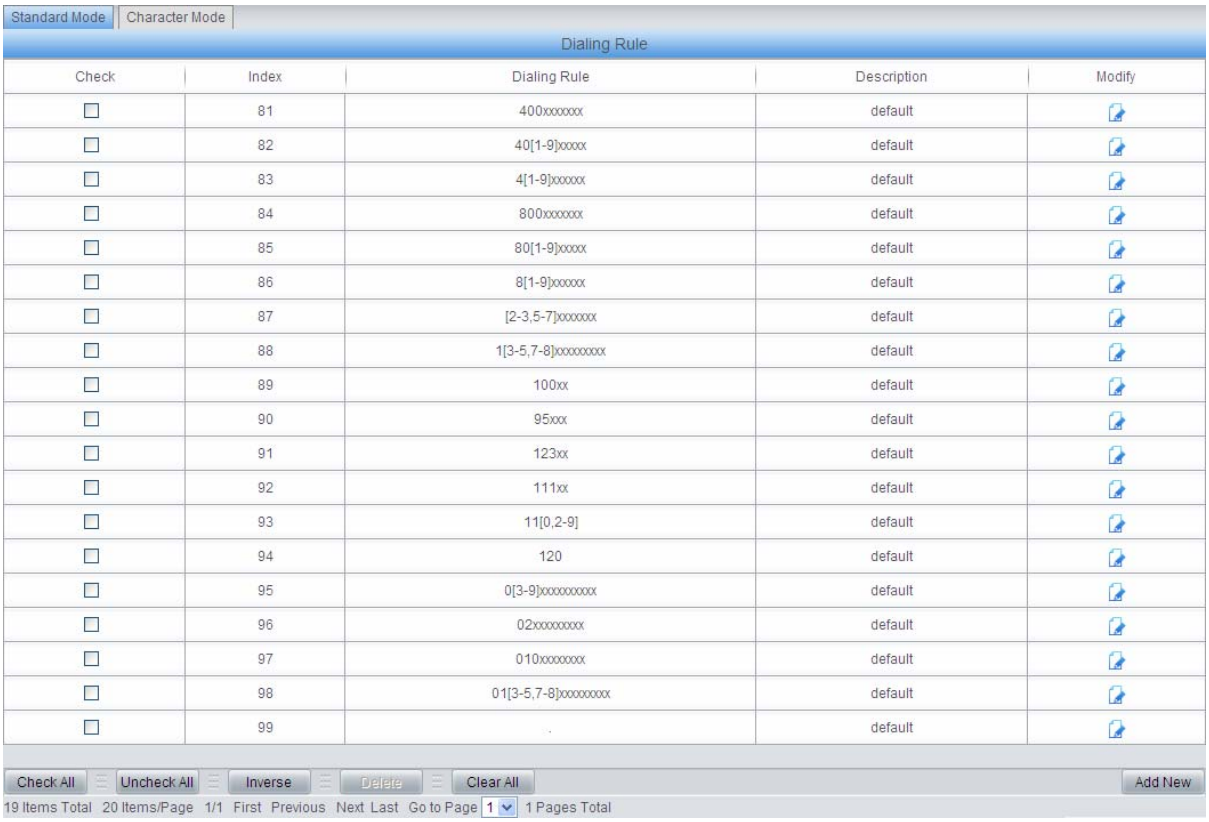

Figure 3-29 Dialing Rule Configuration Interface (Standard)

See Figure 3-29 for the Dialing Rule Configuration interface under the standard mode. The list in the above figure shows the dialing rules with their priorities and description, which can be added by the ***Add New*** button on the bottom right corner. See Figure 3-30 for the dialing rule adding interface.

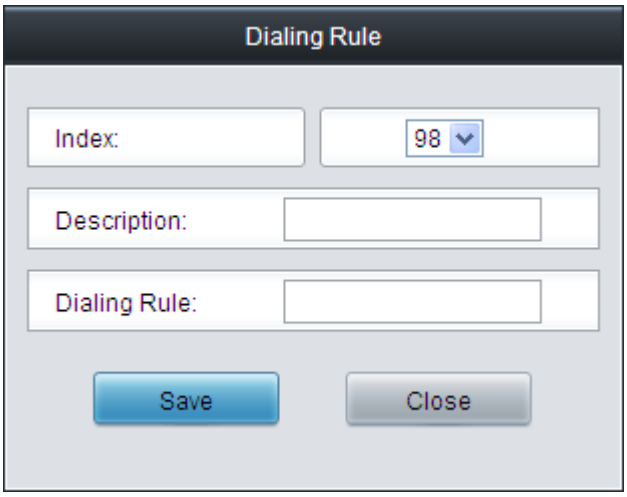

Figure 3-30 Add New Dialing Rule

The table below explains the items shown in Figure 3-30.

Item	Description
Index	The unique index of each dialing rule, which denotes its priority. A dialing rule with a smaller index value has a higher priority and will be checked earlier while matching.
Description	Remarks for the dialing rule. It can be any information, but can not be left empty.
Dialing Rule	Up to 99 dialing rules can be configured in the gateway, and the maximum length of

each dialing rule is 127 characters. See below for the meaning of each character in the dialing rule. The gateway will do instant matching for your dialing number based on the dialing rule and regard your dialing as finished upon receiving '#' or dialing timeout.

Character	Description
"0"~"9"	Digits 0～9.
"A"~"D"	Letters A～D.
"x"	A random number. A string of 'x's represents several random numbers. For example, 'xxx' denotes 3 random numbers.
"."	'.' indicates a random amount (including zero) of characters after it.
"[ ]"	'[ ]' is used to define the range for a number. Values within it only can be digits '0~9', punctuations '-' and ',. For example, [1-3,6,8] indicates any one of the numbers 1, 2, 3, 6, 8.
"-"	'-' is used only in '[ ]' between two numbers to indicates any number between these two numbers.
","	',' is used to separate numbers or number ranges, representing alternatives.
"*"	Only represents symbol "*".
"#"	Only set it at the beginning of the string, representing symbol "#".

There are 19 dialing rules already configured on the gateway for easy use. See below for detailed information.

Priority	Dialing Rule	Description
99	.	Any number in any length.
98	01[3-5,7-8]xxxxxxxxx.	Any 12-digit number starting with 013, 014, 015, 017 or 018
97	010xxxxxxx	Any 11-digit number starting with 010
96	02xxxxxxxx	Any 11-digit number starting with 02
95	0[3-9]xxxxxxxxx	Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09
94	120	Number 120。
93	11[0,2-9]	Number 110, 112, 113, 114, 115, 116, 117, 118 or 119
92	111xx	Any 5-digit number starting with 111
91	123xx	Any 5-digit number starting with 123
90	95xxx	Any 5-digit number starting with 95
89	100xx	Any 5-digit number starting with 100
88	1[3-5,7-8]xxxxxxxxx	Any 11-digit number starting with 13, 14, 15, 17 or 18
87	[2-3,5-7]xxxxxxx	Any 8-digit number starting with 2, 3, 5, 6 or 7

	86	8[1-9]xxxxxx	Any 8-digit number starting with 81, 82, 83, 84, 85, 86, 87, 88 or 89
	85	80[1-9]xxxxx	Any 8-digit number starting with 801, 802, 803, 804, 805,.806, 807, 808 or 809
	84	800xxxxxxx	Any 10-digit number starting with 800
	83	4[1-9]xxxxxx	Any 8-digit number starting with 41, 42, 43, 44, 45, 46, 47, 48 or 49.
	82	40[1-9]xxxxx	Any 8-digit number starting with 401, 402, 403, 404, 405, 406, 407, 408 or 409
	81	400xxxxxxx	Any 10-digit number starting with 400

After configuration, click ***Save*** to save the above settings into the gateway or click ***Close*** to cancel the settings.

Click ***Modify*** in Figure 3-29 to modify the dialing rules. See Figure 3-31 for the dialing rule modification interface. The configuration items on this interface are the same as those on the ***Add New Dialing Rule*** interface.

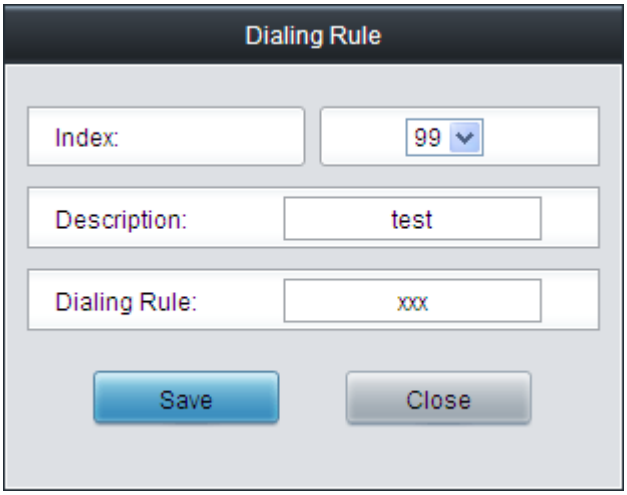

Figure 3-31 Modify Dialing Rule

To delete a dialing rule, check the checkbox before the corresponding index in Figure 3-29 and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all dialing rules at a time, click the ***Clear All*** button in Figure 3-29.

See Figure 3-32 for the Dialing Rule Configuration interface under the Character mode. You can edit the dialing rule list to add a new one or modify an old one. The exact meaning of each rule element is described on the page.

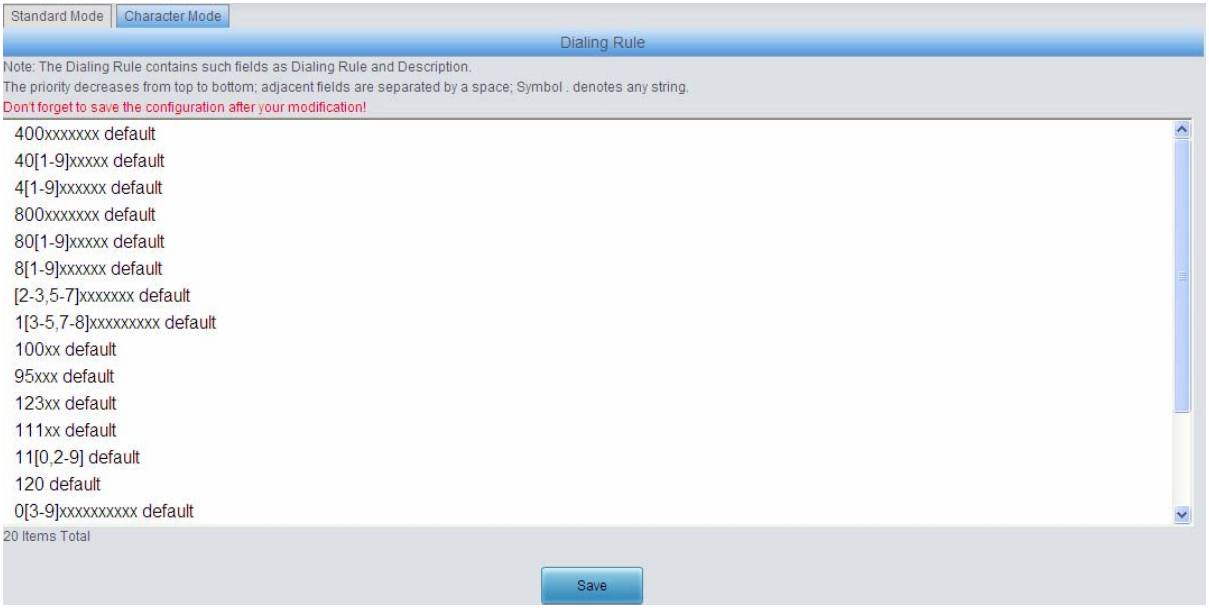

Figure 3-32 Dialing Rule Configuration Interface (Character)

3.5.5 Function Key

See Figure 3-33 for the function key configuration interface. Here you can set a cluster of combination keys to query a related number.

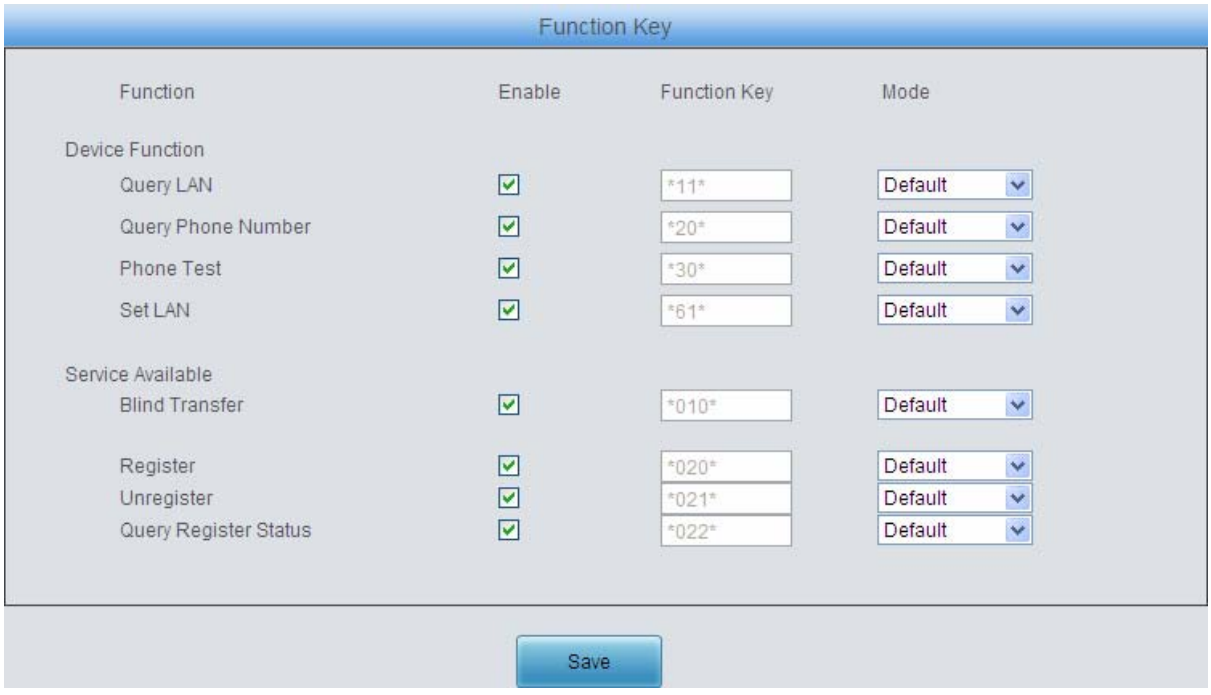

Figure 3-33 Function Key Configuration Interface

Click “Enable” to enable the corresponding function key. The gateway will use the default function keys when the mode is set to default; and it will allow you to set new function keys when the mode is set to user-defined. Click ***Save*** to save your settings into the gateway.

3.5.6 Cue Tone

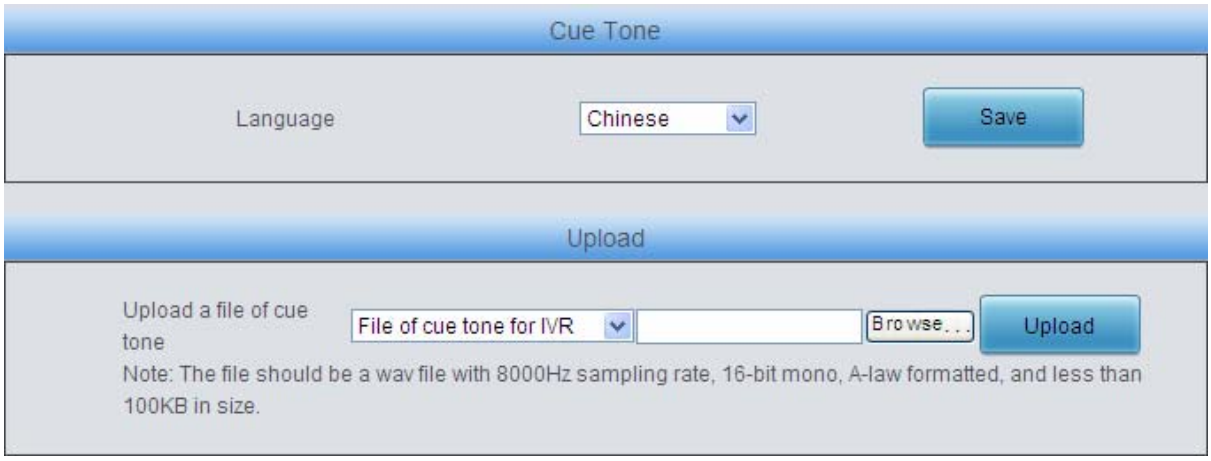
Cue Tone
Language
Chinese
Save
Upload
Upload a file of cue tone
File of cue tone for IVR
Browse...
Upload
Note: The file should be a wav file with 8000Hz sampling rate, 16-bit mono, A-law formatted, and less than 100KB in size.

Figure 3-34 Cue Tone Interface

See Figure 3-34 for the Cue Tone interface. The table below explains the items shown in the above figure.

Item	Description
Language	Sets the language for the gateway to play voice, including two options Chinese and English. The default setting is *Chinese*.
Upload a file of cue tone	Uploads a user-defined cue tone file to the gateway.

Click ***Save*** to save the above settings into the gateway.

3.5.7 QoS

Figure 3-35 Differentiated Services Setting Interface

See Figure 3-35 for the Differentiated Services setting interface. Using this technology, the gateway can meet various application requirements under a limited bandwidth and ensure neither delay nor discard for important services so as to improve its quality of services.

The table below explains the items shown in the above figure.

Item	Description
QoS	Sets whether to enable the OoS differentiated services. By default, it is disabled.
Media Premium QoS	Sets the priority of the media premium for QoS. A media premium QoS with a bigger value has a higher priority. The value range is 0~63, with the default value of 46.

Control Premium QoS	Sets the priority of the control premium for QoS. A control premium QoS with a bigger value has a higher priority. The value range is 0~63, with the default value of 26.

3.5.8 Tone Generator

Tone Generator
Tone Energy (dB)
0
Dial Tone
450/1500
Ringback Tone
450/1000, 0/4000
Busy Tone
450/350, 0/350
FreqA/TimeA, FreqB+FreqC/TimeB
Repeatedly play tones in turn: first, TimeA, a single tone with FreqA, then, Time B, a dual tone composed of FreqB and FreqC.
FreqA+FreqB+FreqC/TimeA, FreqD/TimeB
Repeatedly play tones in turn: first, TimeA, a triple tone composed of FreqA, FreqB and FreqC, then, TimeB, a single tone with FreqD.
Note:
The play time is calculated by ms and cannot be larger than 16383ms for each toneunit. A tone is allowed to contain at most 5 different toneunits and 4 different frequencies, but the frequency and duration of the first toneunit cannot be 0. Frequency being 0 means the toneunit is a piece of silence.
Save
Reset

Figure 3-36 Tone Generator Setting Interface

See Figure 3-36 for the Tone Generator Setting interface. By default, there are three tones on it: Dial Tone—a single tone with 450HZ frequency, plays continuously; Ringback Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 1s play and 4s pause; Busy Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 350ms play and 350ms pause. You can configure the tone generator manually. The exact explanation about the format and the meaning is described on the right of the interface. The value range of the tone energy herein above is -12~17, calculated by dB, with the default value of 0.

3.6 Wireless Settings

Wireless Settings includes six parts: ***Basic Param***, ***Wireless Service***, ***Wireless Module***, ***Short Message***, ***IMEI*** and ***USSD***. See Figure 3-37.

Figure 3-37 Wireless Settings

3.6.1 Basic Parameters

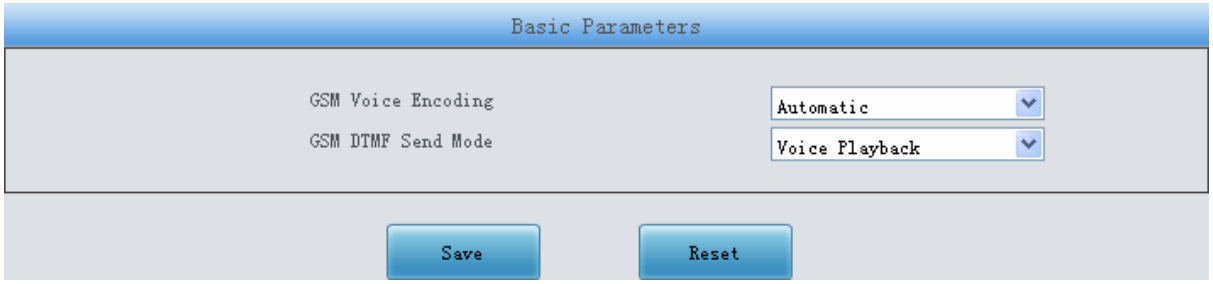

Figure 3-38 Basic Parameters Setting Interface

See Figure 3-38 for the basic parameters setting interface. The table below explains the items shown in the above figure.

Item	Description
GSM Voice Encoding	Sets the mode of the GSM voice encoding. The default value is *Automatic*.
GSM DTMF Send Mode	Sets the mode to send the GSM DTMF, two options including: Voice Playback and Remote Transmission. The default value is *Voice Playback*.

Click Save to save the setting into the gateway, click Reset to restore the configurations.

3.6.2 Wireless Service

Port	FWD Unconditionally	FWD on Busy	FWD on No Reply	FWD on Unreachable	Set Call FWD	Modify
1	15932856225	Close	Close	Close	Successful	
2	Close	Close	Close	Close	---	--
3	Close	Close	Close	Close	---	--
4	Close	Close	Close	Close	---	--
5	Close	Close	Close	Close	---	--
6	Close	Close	Close	Close	---	--
7	Close	Close	Close	Close	---	--
8	Close	Close	Close	Close	---	--

Figure 3-39 Wireless Service Configuration Interface

See Figure 3-39 for the Wireless Service Configuration interface. The table below explains the items shown in the above figure.

Item	Description
FWD Unconditionally	Sets whether to enable the feature of FWD unconditionally and the FWD number if it is enabled.
FWD on Busy	Sets whether to enable the feature of FWD on busy and the FWD number if it is enabled.
FWD on No Reply	Sets whether to enable the feature of FWD on no reply and the FWD number if it is enabled.
FWD on Unreachable	Sets whether to enable the feature of FWD on unreachable and the FWD number if it is enabled.
Set Call Forward	Display the result of the call forward settings.

Cancel All	Cancels all the setting on call FWD service. This item will appear if none of the call FWD is selected.

Click ***Modify*** in Figure 3-39 to modify the properties of the corresponding port. See Figure 3-40 for the Wireless Service Modification interface. Then click ***Modify*** to save the settings into the gateway. It will take some time to apply the settings, and you can check the result in the 'Set Call FWD' column. Click ***Reset*** to restore the configurations, or click ***Cancel*** to cancel the settings.

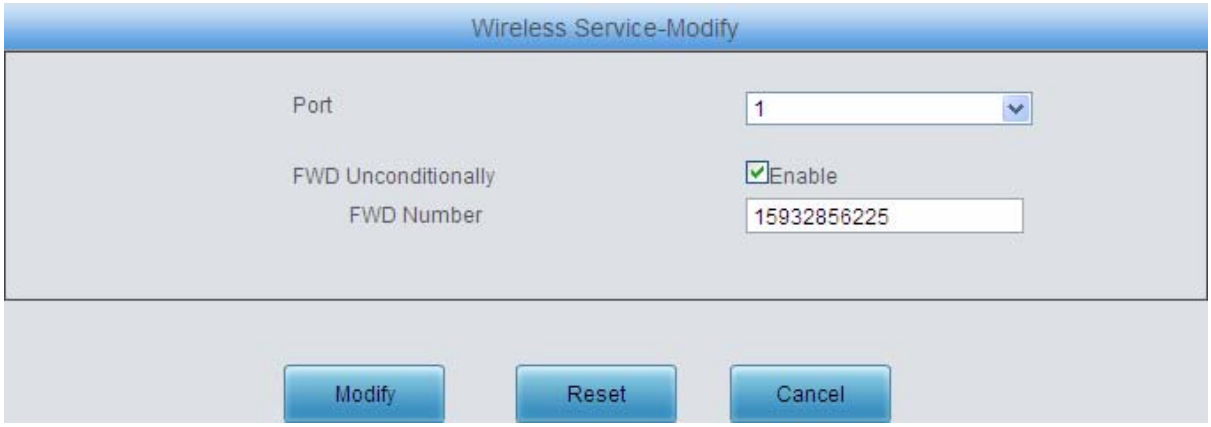

Figure 3-40 Wireless Service Modification Interface

3.6.3 Wireless Module

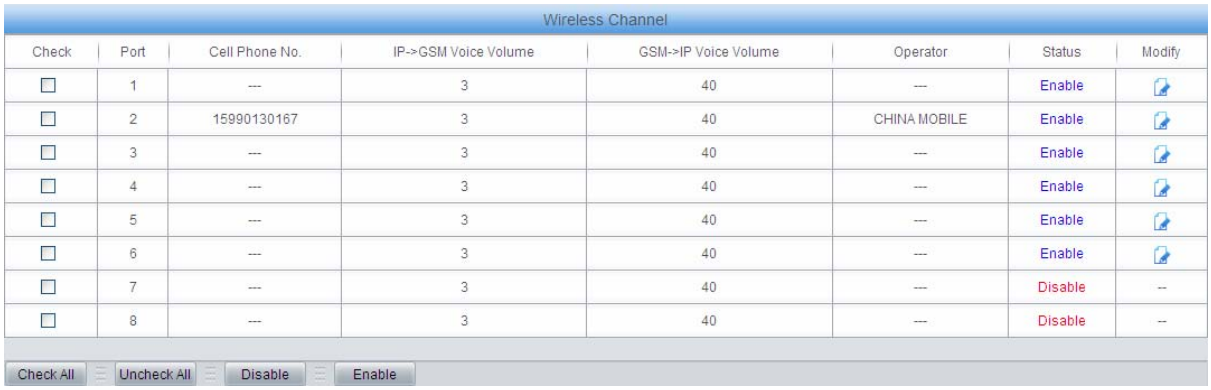

Figure 3-41 Wireless Module Configuration Interface

See Figure 3-41 for the Wireless Module Configuration interface. Click ***Modify*** in Figure 3-41 to modify the properties of the corresponding module. See Figure 3-42 for the Wireless Module Modification interface.

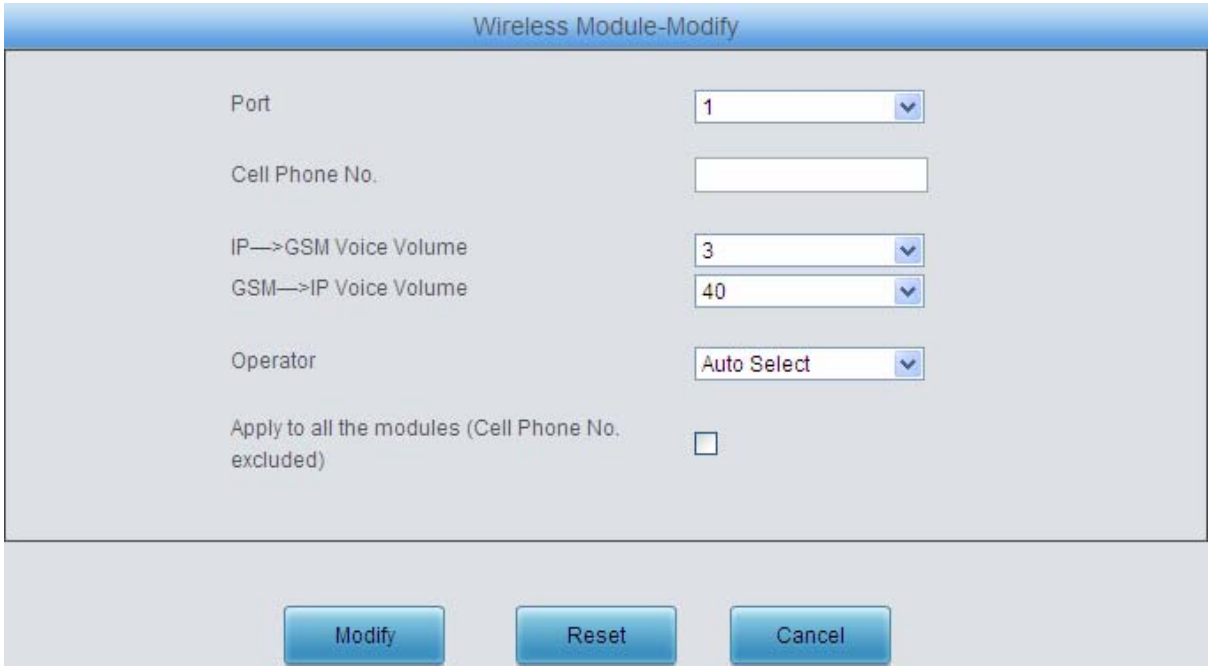

Figure 3-42 Wireless Module Modification Interface

The table below explains the configuration items on the Wireless Module Modification interface.

Item	Description
Port	Serial number of the wireless port on the device.
Cell Phone No.	The number of the SIM card corresponds to the wireless module. This number should be added manually.
IP->GSM Voice Volume	The voice volume from IP to GSM.
GSM->IP Voice Volume	The voice volume from GSM to IP.
Operator	The operator of the wireless module. It is obtained automatically.
Apply to all the modules	Sets whether to apply all the settings except for the cell phone No. to all the modules.

Click ***Modify*** to save the setting into the gateway, click ***Reset*** to restore the configurations, or click ***Cancel*** to cancel the settings.

3.6.4 Short Message

Figure 3-43 Short Message Interface

See Figure 3-43 for the Short Message interface which displays the related information about the received/sent SMS.

Click ***SMS Center*** to go into the SMS Center Modification interface. See Figure 3-44. Click ***Save*** to save the settings into the gateway, click ***Close*** to cancel the settings.

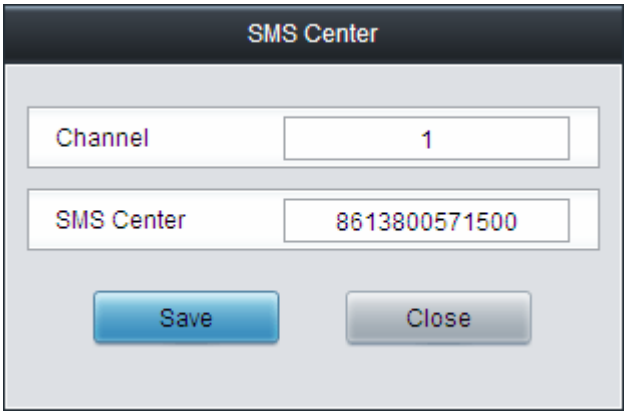

Figure 3-44 SMS Center Modification Interface

Click ***SMS Receiver Details*** in Figure 3-43 to go into the SMS Receiver Details interface. See Figure 3-45. Such information as the remote cell phone number, the time and the content will be displayed on this page.

Figure 3-45 SMS Receiver Details Interface

To delete a piece of SMS receiver detail, check the checkbox before the corresponding index in Figure 3-45 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; to clear all SMS receiver details at a time, click the ***Clear All*** button in Figure 3-45; to go back to the previous page, click ***Return***.

Click ***Records*** in Figure 3-43 to go into the SMS Records interface. See Figure 3-46. Such information as the receive/send status of the SMS, the remote cell phone number, the time, and the content will be displayed on this page.

Figure 3-46 SMS Records Interface

To delete a piece of record, check the checkbox before the corresponding index in Figure 3-46 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; to clear all records at a time, click the ***Clear All*** button in Figure 3-45; to go back to the previous page, click ***Return***.

Click ***Send SMS*** in Figure 3-43 to go into the Send SMS interface. See Figure 3-47.

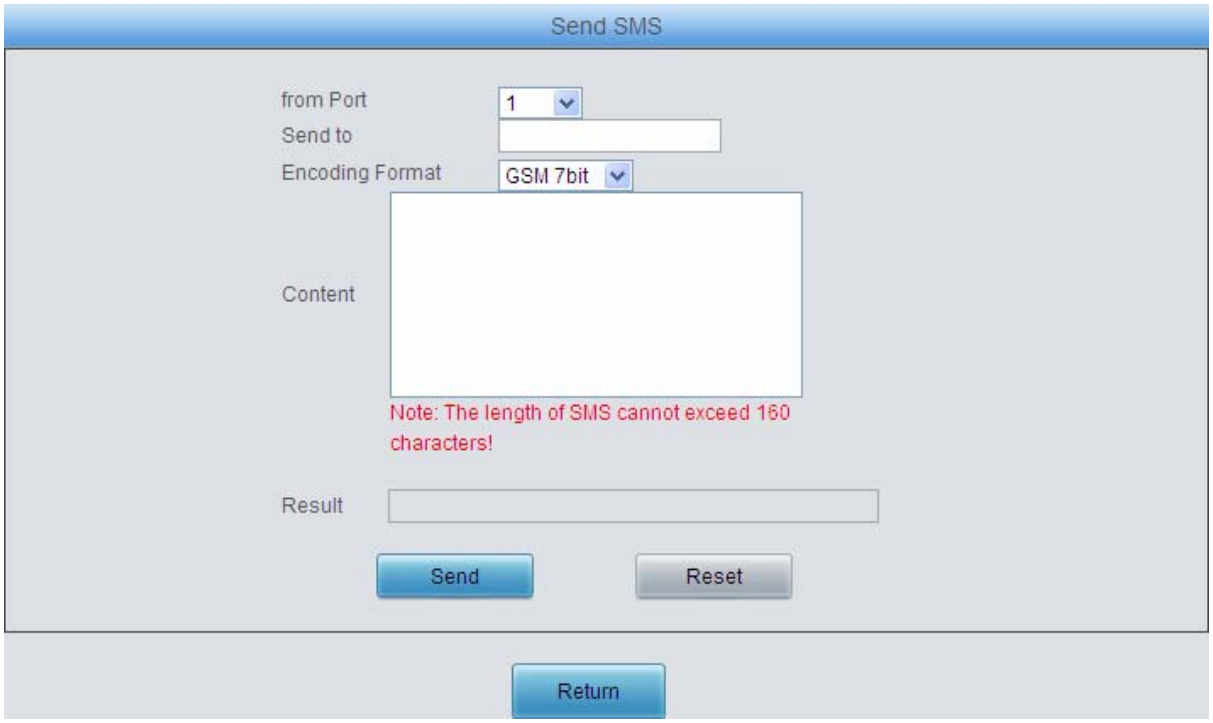

Figure 3-47 Send SMS Interface

The table below explains the configuration items on the Send SMS interface.

Item	Description
from Port	Select a port to send the SMS.
Send to	Enter the remote number to receive the SMS.
Encoding Format	The encoding format for the SMS, two options available: GSM 7bit and UCS2.
Content	The content of the SMS required to be sent.
Result	Display the send result of the SMS.

Click ***Send*** to send out the SMS. Click ***Reset*** to restore the configurations, or click ***Return*** to go back to the previous

3.6.5 IMEI

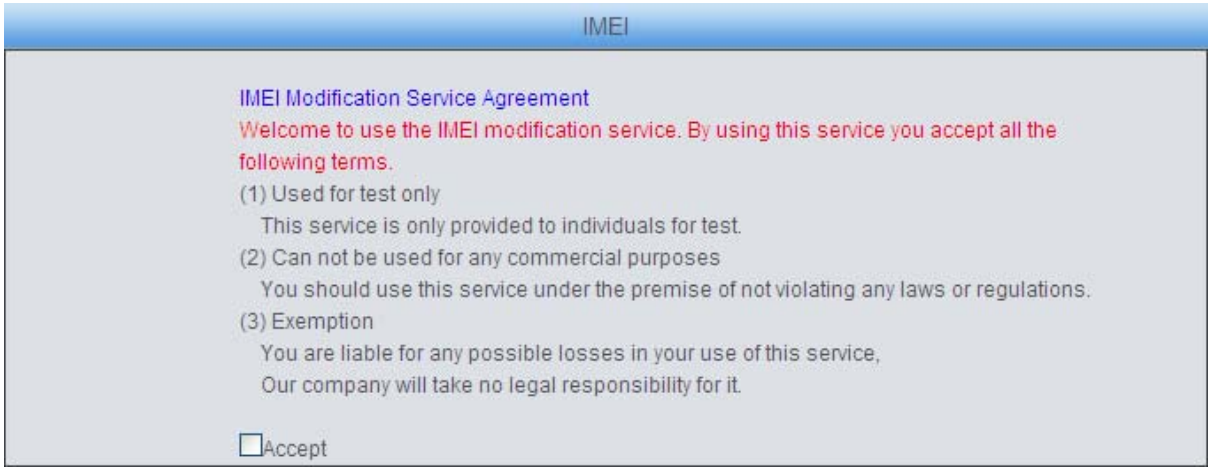
IMEI
IMEI Modification Service Agreement
Welcome to use the IMEI modification service. By using this service you accept all the following terms.
(1) Used for test only
This service is only provided to individuals for test.
(2) Can not be used for any commercial purposes
You should use this service under the premise of not violating any laws or regulations.
(3) Exemption
You are liable for any possible losses in your use of this service,
Our company will take no legal responsibility for it.
Accept

Figure 3-48 IMEI Interface

See Figure 3-48 for the IMEI interface. Read the agreement carefully and click ***Accept*** before you

go into the IMEI Modification interface (Figure 3-49).

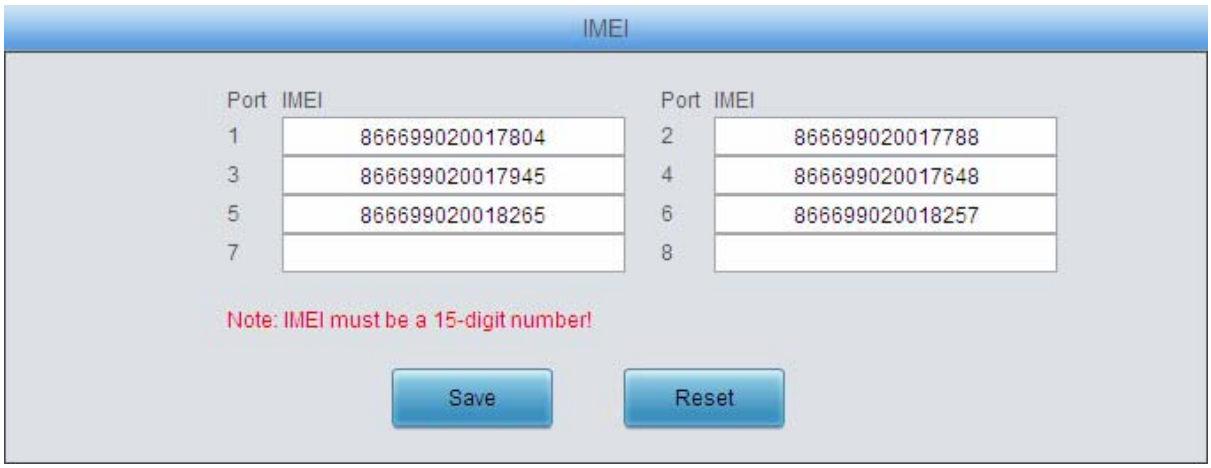

Figure 3-49 IMEI Modification Interface

After configuration, click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations.

3.6.6 USSD

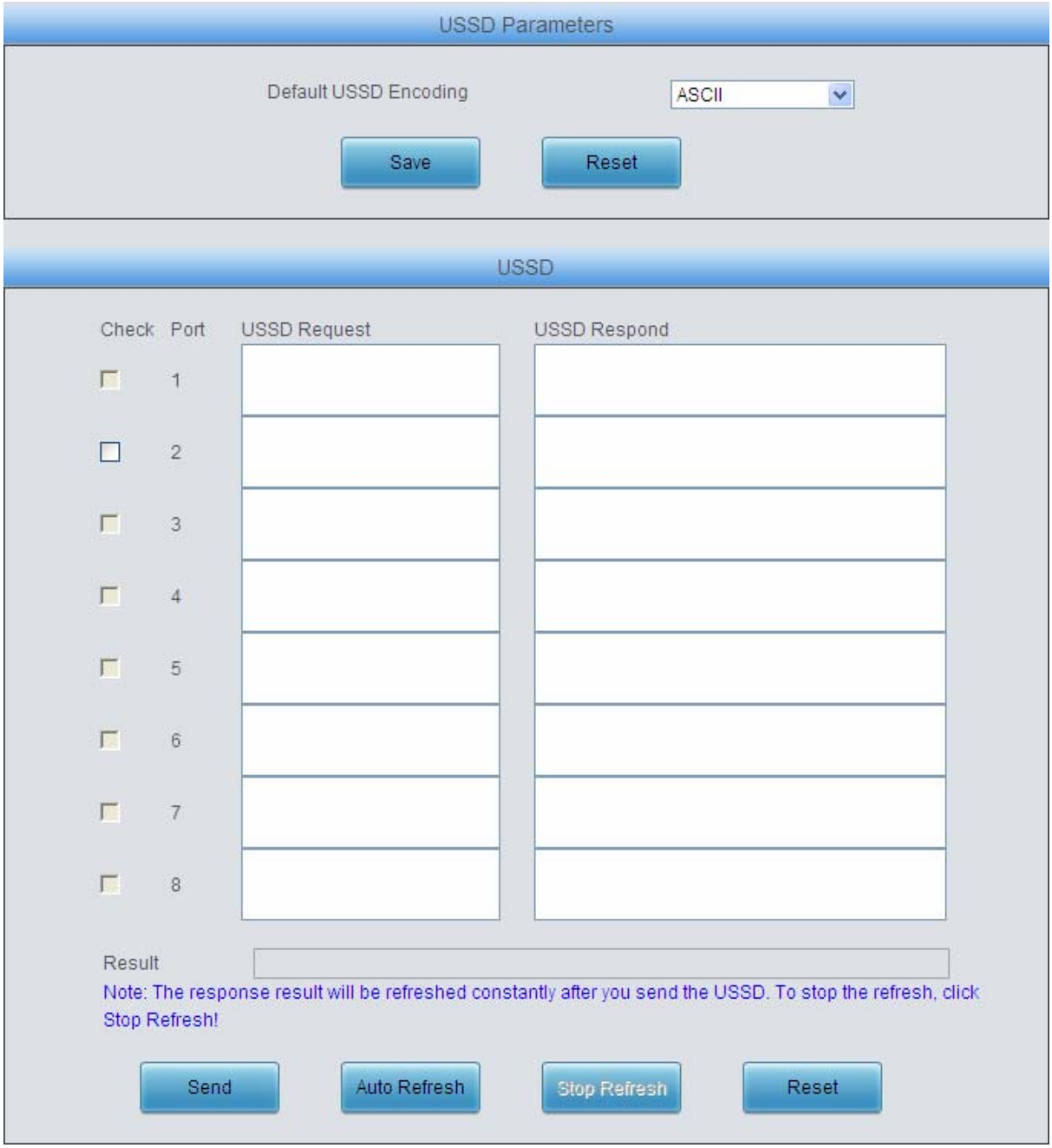

Figure 3-50 USSD Setting Interface

See Figure 3-50 for the USSD Setting interface. The table below explains the items shown in the above figure.

Item	Description
Default USSD Encoding	Sets the default encoding format for USSD, two options available: ASCII and UCS2.
Port	Sets the port used to send the USSD request.
USSD Request	Inputs the content of the USSD request.
USSD Respond	Displays the result of the USSD respond.
All	Selects all the ports to send the same USSD request.
Result	Displays the sent result of the USSD request

Click ***Send*** in Figure 3-50 to send out the USSD request. Click ***Auto Refresh*** to refresh the result automatically and click ***Stop Refresh*** to stop refreshing the result. Click ***Reset*** to restore the configurations.

3.7 Port Settings

Port Settings includes two parts: ***Port*** and ***Port Group***. See Figure 3-51.

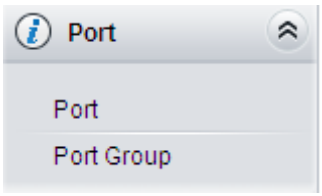

Figure 3-51 Port Settings

3.7.1 Port

Port Settings — Batch Modify

Port	Type	SIP Account	Authentication Username	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Modify
1	GSM	8001	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Disable	
2	GSM	8002	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Disable	
3	GSM	8003	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Disable	
4	GSM	8004	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Disable	
5	GSM	8005	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Disable	
6	GSM	8006	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Disable	
7	GSM	8007	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Disable	
8	GSM	8008	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Disable	

Figure 3-52 Port Settings Interface

See Figure 3-52 for the Port Settings interface. The list in the above figure shows the feature and properties of each port. Click ***Modify*** in Figure 3-52 to modify the properties of the corresponding port. See Figure 3-53 for the Port Modification interface.

Figure 3-53 Port Modification

The table below explains the configuration items on the port modification interface.

<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>Port</td><td>Serial number of the port on the device.</td></tr>
<tr><td>Register Port</td><td>Sets whether to register the port to the SIP server.
When this item is set to No, the item Reg Status on the Port Settings interface (Figure 3-52) shows Unregistered; when this item is set to Yes, the item Reg Status shows Failed or Registered.</td></tr>
<tr><td>SIP Account</td><td>When the port initiates a call to SIP, this item corresponds to the username of SIP. The default SIP account is 80XX among which XX represents the corresponding port number. For example, the default SIP account corresponding to Port 1 is 8001, and that corresponding to Port 8 is 8008.</td></tr>
<tr><td>Password</td><td>Registration password of the port. To register a port to the SIP server, both items SIP Account and Password must be filled in.</td></tr>
<tr><td>Authentication Username</td><td>Authentication username of a port, used to register the port to the SIP server when IMS network is enabled.
Note: This item appears only when IMS Network is enabled.</td></tr>
<tr><td>Connection Method</td><td>Port connection methods include:
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Static Binding</td><td>Bind the number to a wireless port. The number will be listed in the Bound Number column.</td></tr>
<tr><td>Two Stages Dialing Mode (default)</td><td>Under this mode, an incoming call from a wireless port will go into the IVR system. Then IVR will play a speech prompt “Please dial the extension number”. If you fail to input the correct target number before IVR finishes the third repeat of the prompt, the port will hang up the call automatically; otherwise, the call goes out successfully.</td></tr>
</table>
Note: Both items Connection Method and Bound Number will be hidden if the SIP Station feature is enabled on the SIP Settings interface.</td></tr>
<tr><td>Echo Canceller</td><td>The echo cancellation feature for a call conversation over the wireless channel. By default, this feature is disabled and the effect can reach 128ms.</td></tr>
<tr><td>Forbid Outgoing Call</td><td>If this feature is enabled, the port will be forbidden to call out. The default setting is disabled.</td></tr>
<tr><td>Caller ID Detection</td><td>If this feature is enabled, the port will detect the Caller IDs from the incoming calls. The default setting is disabled.</td></tr>
</table>

After configuration, click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Cancel*** to cancel the settings.

Or you can click ***Batch*** to modify several pieces of port settings at the same time. See Figure 3-54 below for the Port Batch Modification interface.

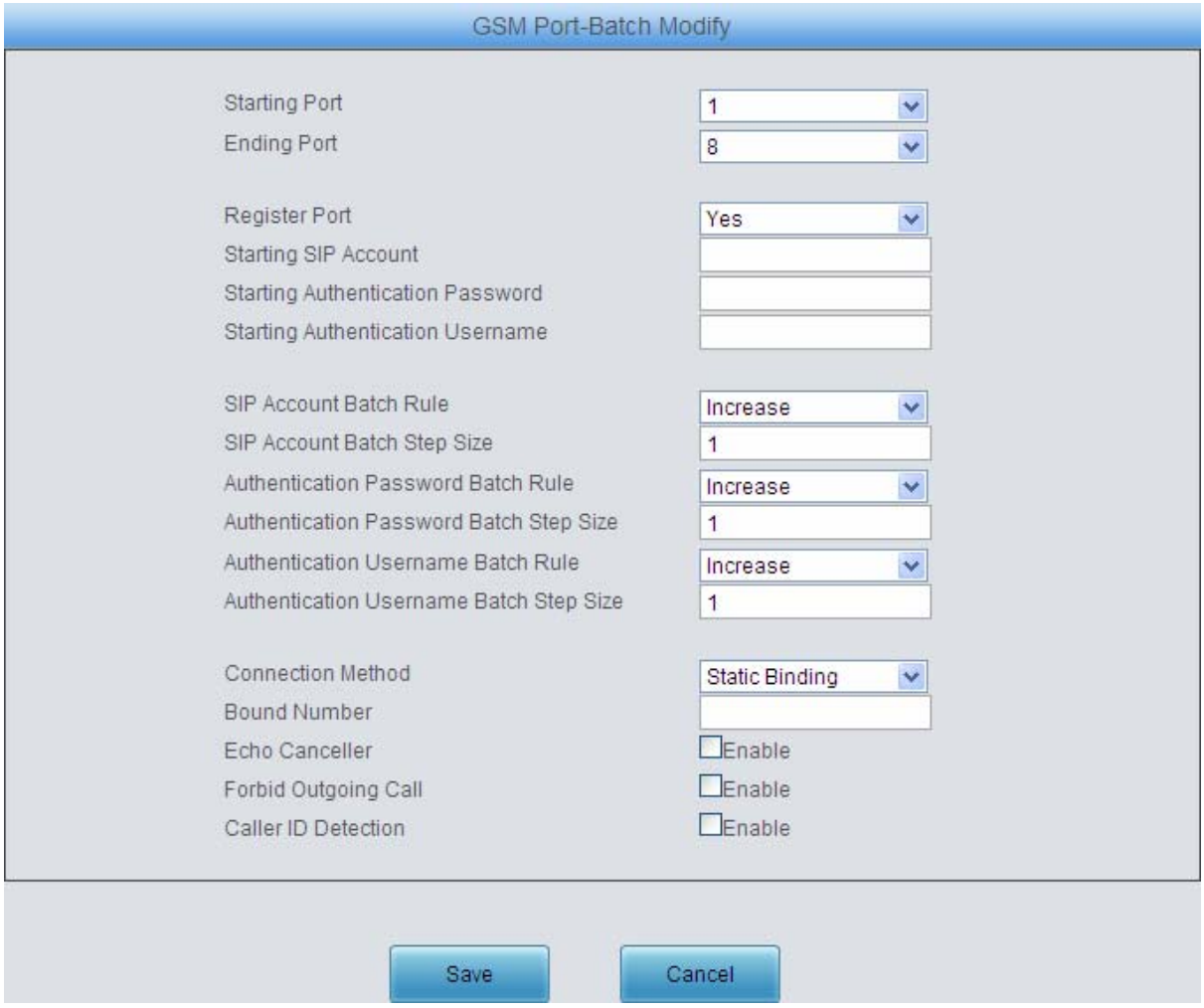

Figure 3-54 Port Batch Modification

Some configuration items on this interface are the same as those on the ***Port Modification Interface***. The others are described in the table below.

Item	Description
Starting Port	The starting serial number of the port on the device in the batch setting.
Ending Port	The ending serial number of the port on the device in the batch setting.
Starting SIP Account	The starting SIP account in the batch setting.
Starting Authentication Password	The starting authentication password in the batch setting.
Starting Authentication Username	The starting authentication username in the batch setting.
SIP Account Batch Rule	The rule for batch setting the SIP account, including ***Increase*** and ***Decrease*** two options.
SIP Account Batch Step Size	Sets the increase or decrease step size of the SIP account in the batch setting.
Authentication Password Batch Rule	The rule for batch setting the authentication password, including ***Increase*** and ***Decrease*** two options.
Authentication Password Batch Step Size	Sets the increase or decrease step size of the authentication password in the batch setting.

Authentication Username Batch Rule	The rule for batch setting the authentication username, including ***Increase*** and ***Decrease*** two options.
Authentication Username Batch Step Size	Sets the increase or decrease step size of the authentication username in the batch setting.

After configuration, click ***Save*** to save the settings into the gateway, or click ***Cancel*** to cancel the settings.

3.7.2 Port Group

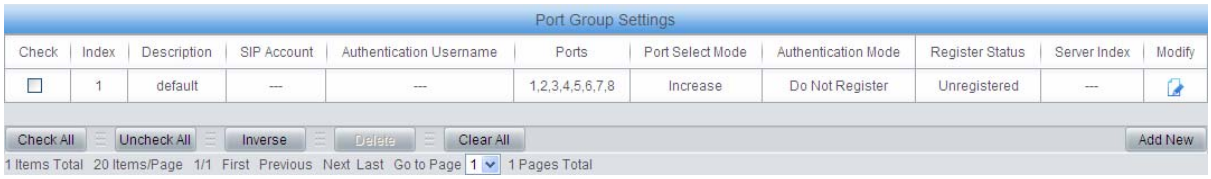

Figure 3-55 Port Group Settings Interface

See Figure 3-55 for the port group settings interface. A port group is a set containing single or multiple ports, used to specify such properties as ***Port Selection*** and ***Authentication Mode*** for all the ports in it. A new port group can be added by the ***Add New*** button on the bottom right corner of the above list. See Figure 3-56 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.

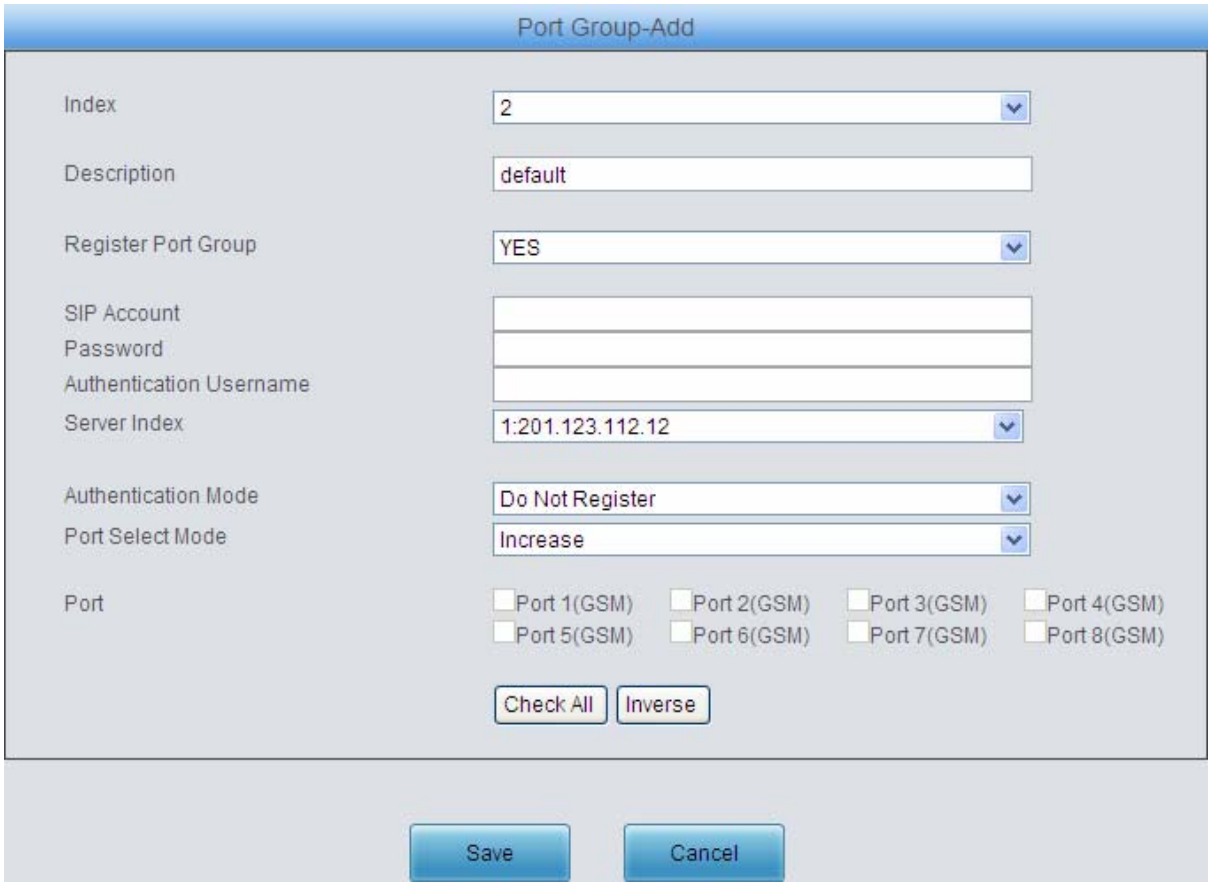

Figure 3-56 Add New Port Group

The table below explains the items in the above figure.

Item	Description

<table>
<tr><td>Index</td><td>The unique index of each port group, which is mainly used in the configuration of routing rules and number manipulation rules to correspond to port groups.</td></tr>
<tr><td>Description</td><td>More information about each port group, with default value of default.</td></tr>
<tr><td>Register Port Group</td><td>To register the port group to the SIP server. Only when this configuration item is set to Yes can you see the configuration items SIP Account and Password.</td></tr>
<tr><td>SIP Account</td><td>When the port group initiates a call to SIP, this item corresponds to the username of SIP.</td></tr>
<tr><td>Password</td><td>Registration password of the port group. To register the port group to the SIP server, both configuration items SIP Account and Password should be filled in.</td></tr>
<tr><td>Authentication Username</td><td>Authentication username of a port, used to register the port to the SIP server when IMS network is enabled.
Note: This item appears only when IMS Network is enabled.</td></tr>
<tr><td>Server Index</td><td>The index of the sip server which will be quoted by the current port.</td></tr>
<tr><td>Authentication Mode</td><td>Sets the way for SIP to make outgoing calls (Tel→IP) on the gateway.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Do Not Register (default)</td><td>SIP initiates a call in a point-to-point mode.</td></tr>
<tr><td>Register Gateway</td><td>SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to 3.4.1 SIP for gateway registration.)</td></tr>
<tr><td>Register Port Group</td><td>SIP initiates a call with the registered SIP account and password of the port group.</td></tr>
<tr><td>Register Port</td><td>SIP initiates a call with the registered SIP account and password of the port.</td></tr>
</table></td></tr>
<tr><td>Register Status</td><td>Registration status of the port group. See Figure 3-55. When Register Port Group is set to No, the value of this item is Unregistered; when Register Port Group is set to Yes, the value of this item may be Failed or Registered.</td></tr>
<tr><td>Port</td><td>The ports in the port group. If the checkbox before a port is grey, it indicates that the port is not available or has been occupied. All selected ports for a port group will be displayed in the Ports column in Figure 3-55. Note: When a port group contains multiple ports, the automatic call forward feature is invalid.</td></tr>
</table>

After configuration, click ***Save*** to save the settings into the gateway, click ***Cancel*** to cancel the settings. ***Check All*** means to select all available ports on the current page; ***Inverse*** means to uncheck the selected items and check the unselected.

Click ***Modify*** at the end of the list in **Port Group Settings Interface** to modify the properties of a port group. See Figure 3-57 for the Port Group Modification interface. The configuration items on this interface are the same as those on the ***Add New Port Group*** interface.

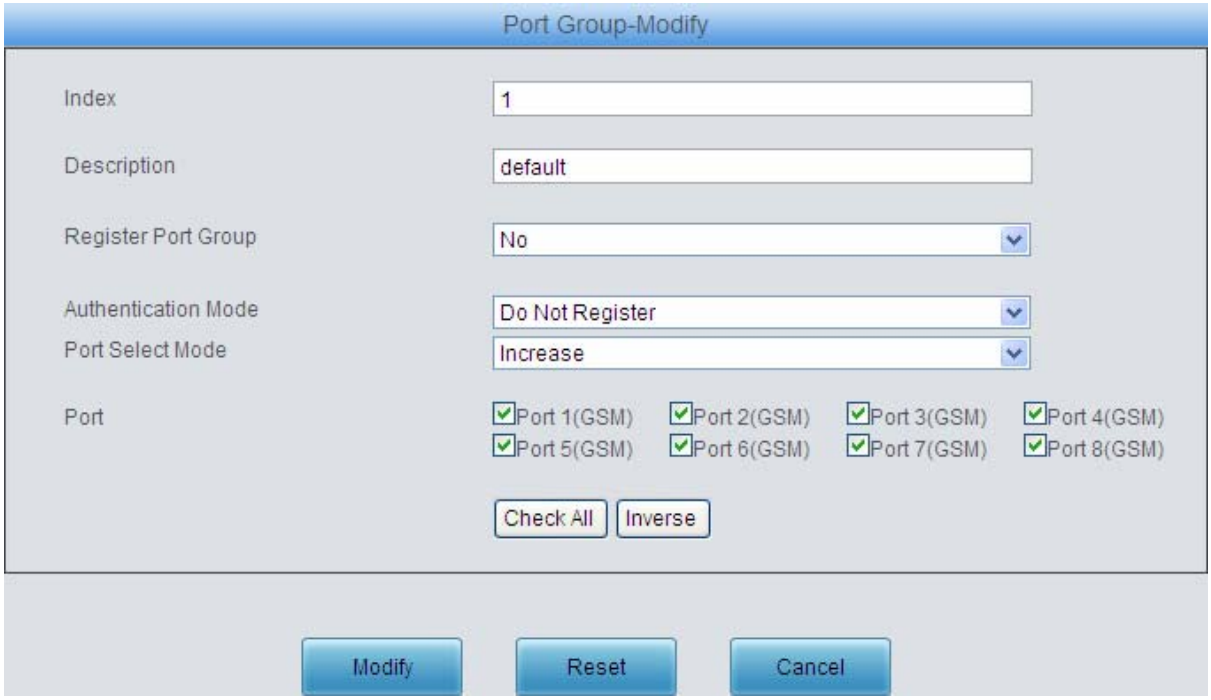

Figure 3-57 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-55 and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all port groups at a time, click the ***Clear All*** button in Figure 3-55.

3.8 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: IP→Tel and Tel→IP. See Figure 3-58.

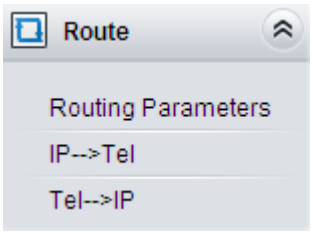

Figure 3-58 Route Settings

3.8.1 Routing Parameters

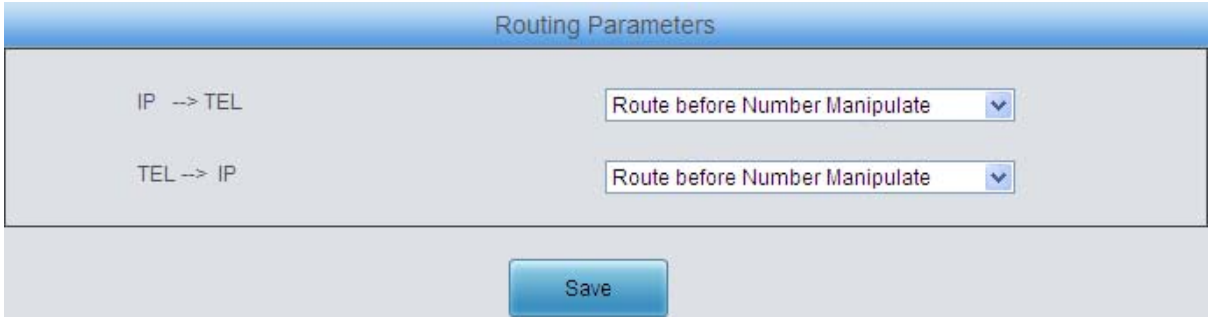

Figure 3-59 Routing Parameters Configuration Interface

See Figure 3-59 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions IP→Tel and Tel→IP to be routing before or after number manipulation. The default value is *Route before Number Manipulate*.

After configuration, click ***Save*** to save the above settings into the gateway.

3.8.2 IP to Tel

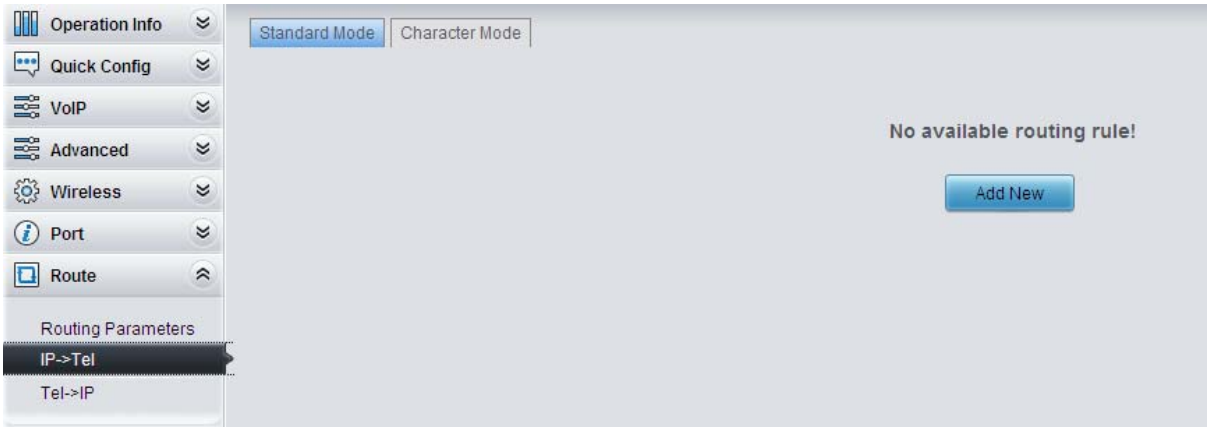

Figure 3-60 IP→Tel Routing Rule Configuration Interface (Standard)

See Figure 3-60 for the IP→Tel routing rule configuration interface. By default, there is no available routing rule on the gateway. The IP→Tel routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click ***Add New*** to add them manually. See Figure 3-61. You may use the default values of all the configuration items herein.

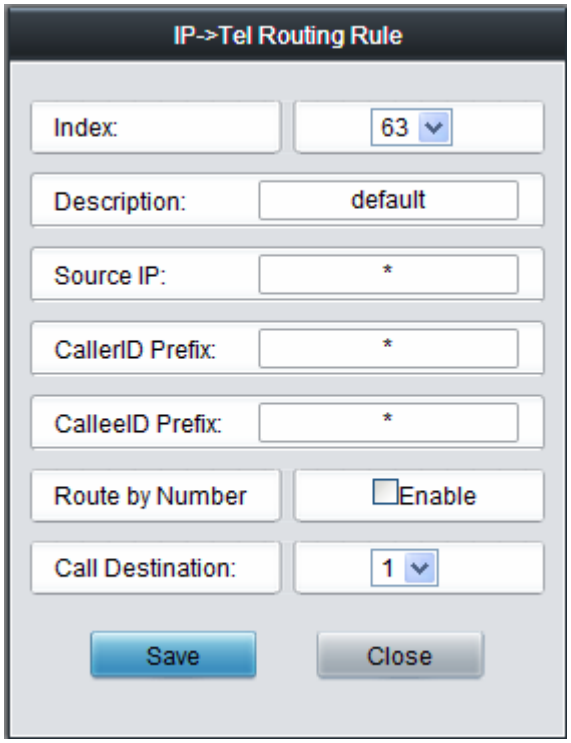

Figure 3-61 Add New Routing Rule (IP→Tel)

The table below explains the items shown in the above figure.

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with

	a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of *default*.
Source IP	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, 、"[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be characters '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicates any character between these two characters. ',' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Source IP*** specify a routing rule for calls. **Note:** "[*]" represents TFM symbol *, while "*" represents any string.
Route by Number	When this feature is enabled, the gateway will route a call from IP to a corresponding port based on its number. And the number of the port which this call will be routed to can be set via the item ***SIP Account*** on the Port Settings interface. In such case, the configuration item ***Call Destination*** goes invalid and shows *Route by Number* on the routing rule configuration interface. The default setting is *disabled*.
Call Destination	Port group to which the call will be routed.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

See Figure 3-62 for the IP→Tel routing rule configuration interface after your configuration. There is a rule displayed with Index 63 and Call Destination 'Route by Number', having no restriction on Source IP, CallerID Prefix and CalleeID Prefix, which indicates the gateway will route a call from any IP address to a corresponding port based on its number.

Press the ***Add New*** button on the bottom right corner of the list to add a new routing rule.

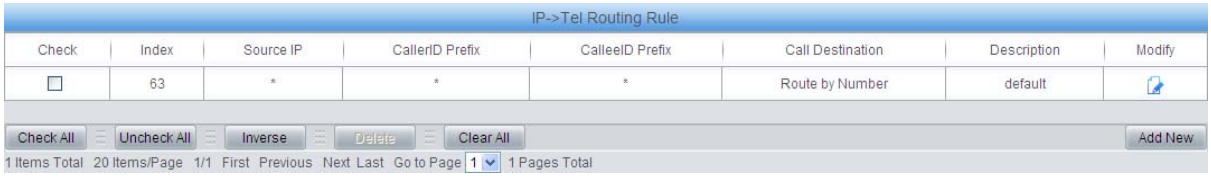

Figure 3-62 IP→Tel Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-62 to modify a routing rule. The configuration items on the IP→Tel routing rule modification interface are the same as those on the ***Add New Routing Rule (IP→Tel)*** interface. Note that the item ***Index*** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-62 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the ***Clear All*** button in Figure 3-62.

See Figure 3-63 for the IP→Tel Routing Rule Configuration Interface under the Character mode. You can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

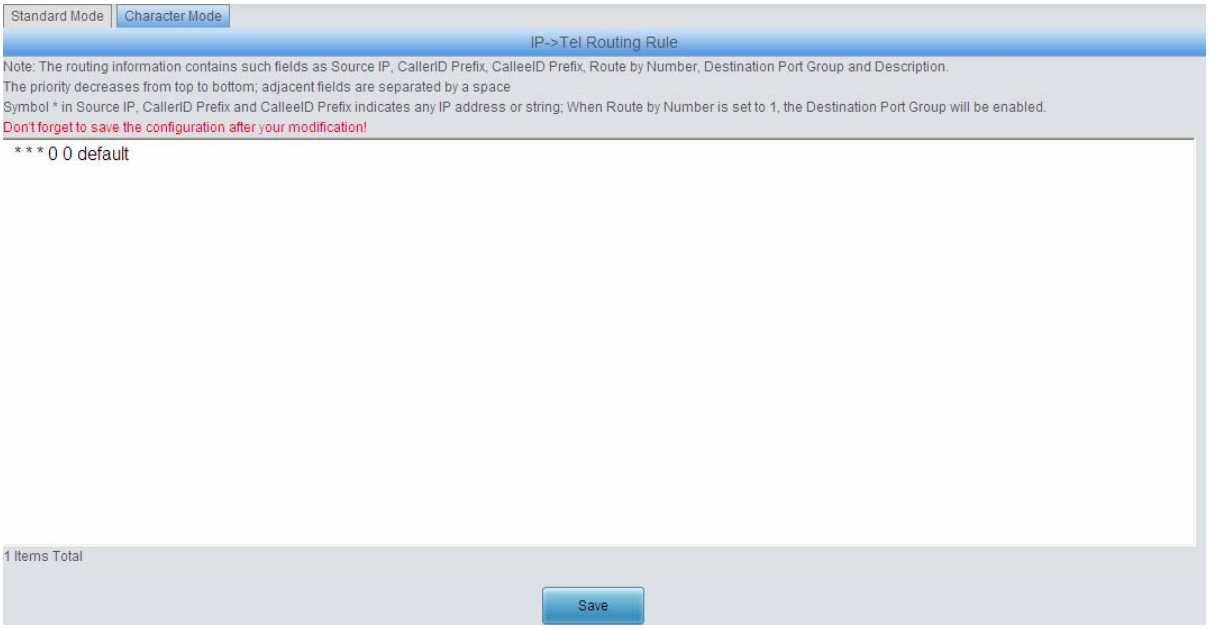
Standard Mode
Character Mode
IP->Tel Routing Rule
Note: The routing information contains such fields as Source IP, CallerID Prefix, CalleeID Prefix, Route by Number, Destination Port Group and Description.
The priority decreases from top to bottom; adjacent fields are separated by a space
Symbol * in Source IP, CallerID Prefix and CalleeID Prefix indicates any IP address or string; When Route by Number is set to 1, the Destination Port Group will be enabled.
Don't forget to save the configuration after your modification!
* * * 0 0 default
1 Items Total
Save

Figure 3-63 IP→Tel Routing Rule Configuration Interface (Character)

3.8.3 Tel to IP

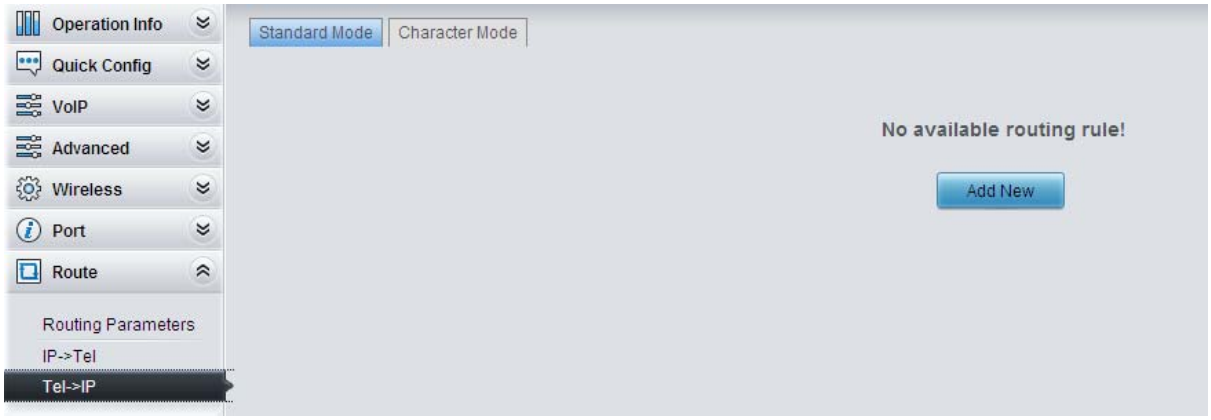

Figure 3-64 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-64 for the Tel→IP routing rule configuration interface. By default, there is no available routing rule on the gateway. The Tel→IP routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click ***Add New*** to add them manually. See Figure 3-65. You may use the default values of all the configuration items herein except for ***Destination IP*** and ***Destination Port***.

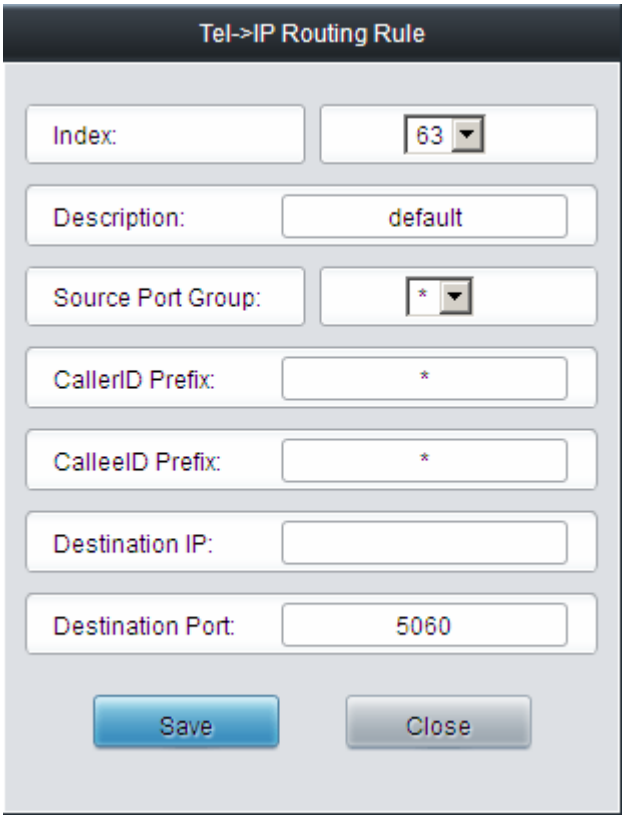

Figure 3-65 Add New Routing Rule (Tel→IP)

The table below explains the items shown in the above figure.

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of *default*.
Source Port Group (Call Initiator)	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or characters ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '**-**' and '**,**'. ('**-**' is used between two characters to indicates any characters between these two characters. '**,**' is used to separate characters or characters ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Source Port Group (Call Initiator)*** specify a routing rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.
Destination IP, Destination Port	IP address and port number of the remote end to which the call will be routed.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

See Figure 3-66 for the Tel→IP routing rule configuration interface after your configuration. There is a rule displayed with Index 63, Destination IP '192.168.1.101' and Destination Port '5060' (i.e. default IP address and port of the gateway), having no restriction on Call Initiator, CallerID Prefix and CalleeID Prefix, which indicates all the outgoing calls from Tel which conform to the dialing rule will be routed to the gateway.

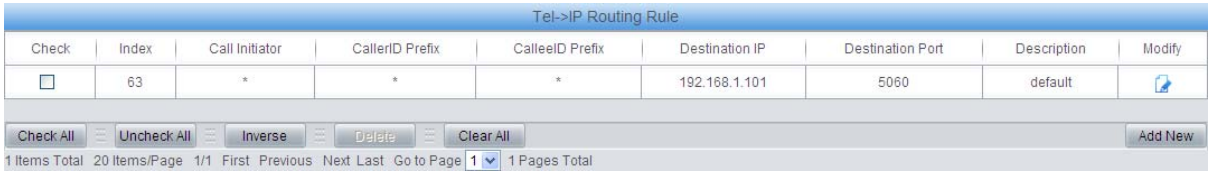

Figure 3-66 Tel→IP Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-66 to modify a routing rule. The configuration items on the Tel→IP routing rule modification interface are the same as those on the ***Add New Routing Rule (Tel→IP)*** interface. Note that the item ***Index*** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-66 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the ***Clear All*** button in Figure 3-66.

See Figure 3-67 for the Tel→IP Routing Rule Configuration Interface under the Character mode. You can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

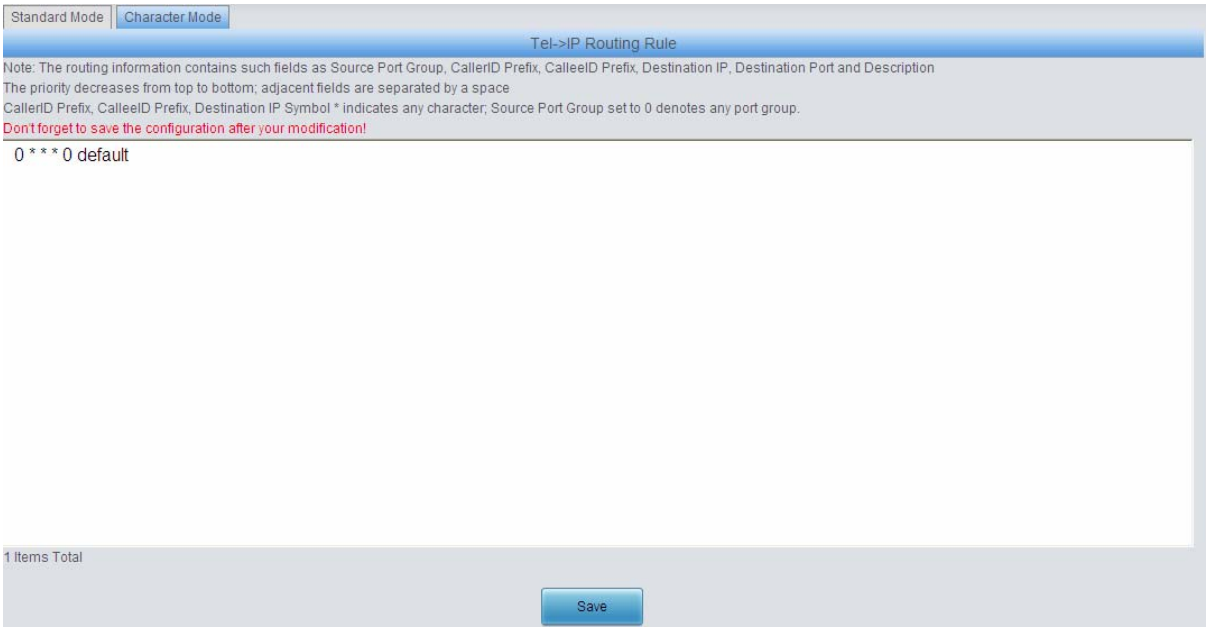
Standard Mode
Character Mode
Tel->IP Routing Rule
Note: The routing information contains such fields as Source Port Group, CallerID Prefix, CalleeID Prefix, Destination IP, Destination Port and Description
The priority decreases from top to bottom; adjacent fields are separated by a space
CallerID Prefix, CalleeID Prefix, Destination IP Symbol * indicates any character; Source Port Group set to 0 denotes any port group.
Don't forget to save the configuration after your modification!
0 * * * 0 default
1 Items Total
Save

Figure 3-67 Tel→IP Routing Rule Configuration Interface (Character)

3.9 Number Manipulation

Number Manipulation includes four parts: **IP→Tel CallerID**, **IP→Tel CalleeID**, **Tel→IP CallerID** and **Tel→IP CalleeID**. See Figure 3-68.

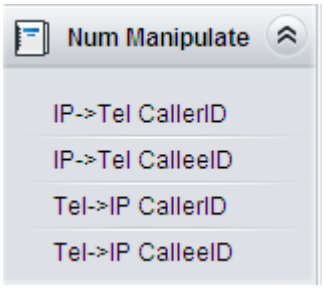

Figure 3-68 Number Manipulation

3.9.1 IP to Tel CallerID

Figure 3-69 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-69 for the IP→Tel CallerID manipulation interface under the Standard mode. A new number manipulation rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-70 for the IP→Tel CallerID manipulation rule adding interface. You may use the default values of all the configuration items herein.

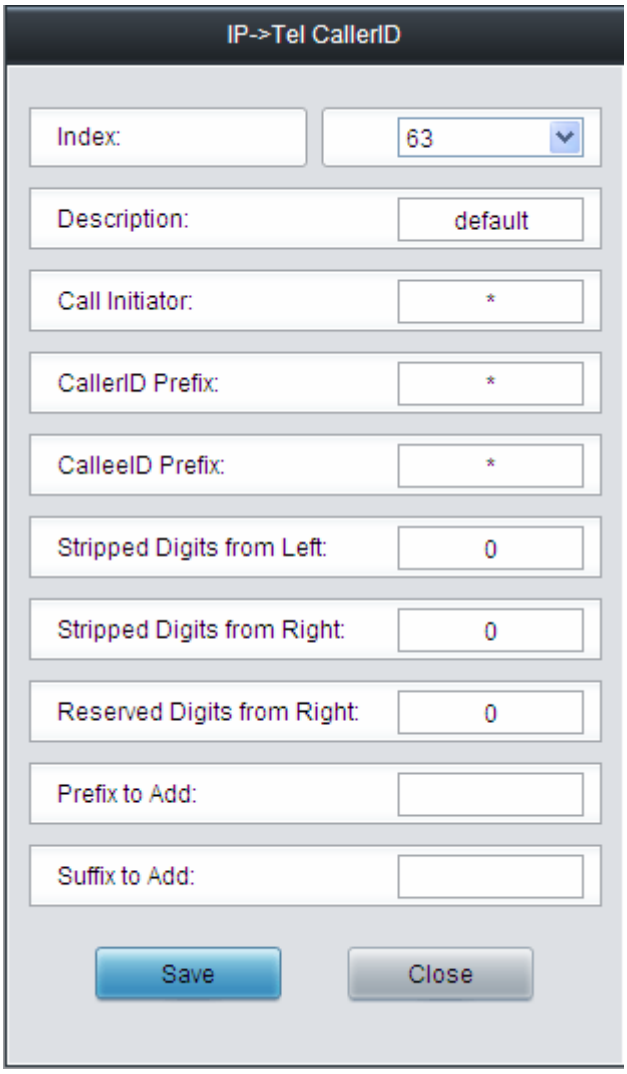

Figure 3-70 Add IP→Tel CallerID Manipulation Rule

The table below explains the items shown in the above figure.

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule, with the default value of *default*.
Call Initiator	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address.

CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '**-**' and '**,**'. ('**-**' is used between two characters to indicates any character between these two characters. '**,**' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Call Initiator*** specify a number manipulation rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

Note: The number manipulation is performed in 5 steps by the order of the following configuration items: *Stripped Digits from Left*, *Stripped Digits from Right*, *Reserved Digits from Right*, *Prefix to Add* and *Suffix to Add*.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

Click ***Modify*** in Figure 3-69 to modify a number manipulation rule. See Figure 3-71 for the IP→Tel CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the ***Add IP→Tel CallerID Manipulation Rule*** interface. Note that the item ***Index*** cannot be modified.

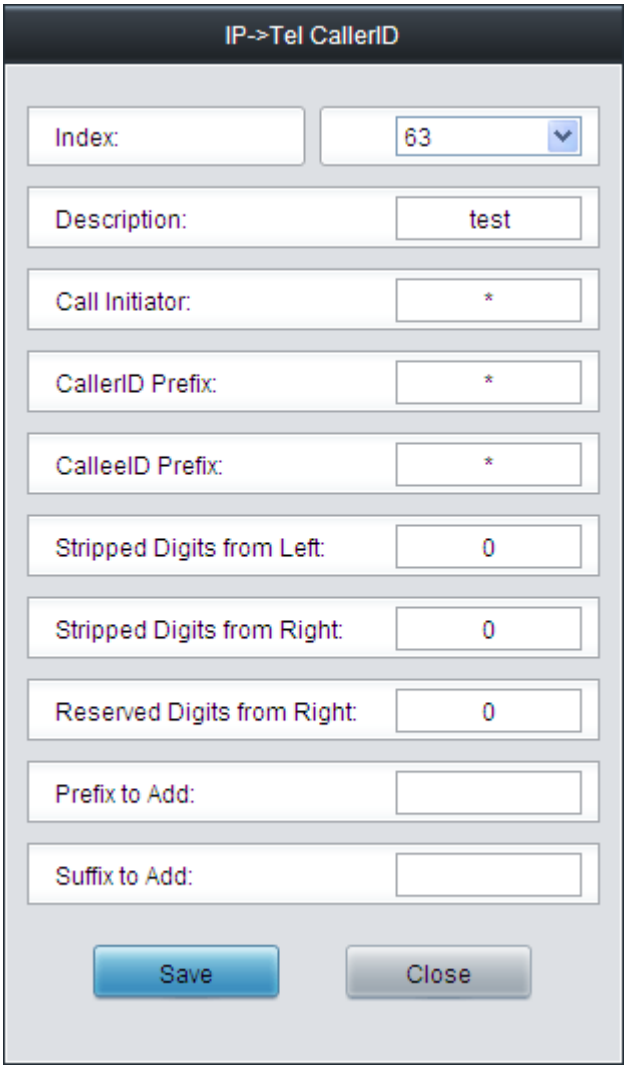

Figure 3-71 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-69 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all number manipulation rules at a time, click the ***Clear All*** button in Figure 3-69.

See Figure 3-72 for the IP→Tel CallerID Manipulation Interface under the Character mode. You can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

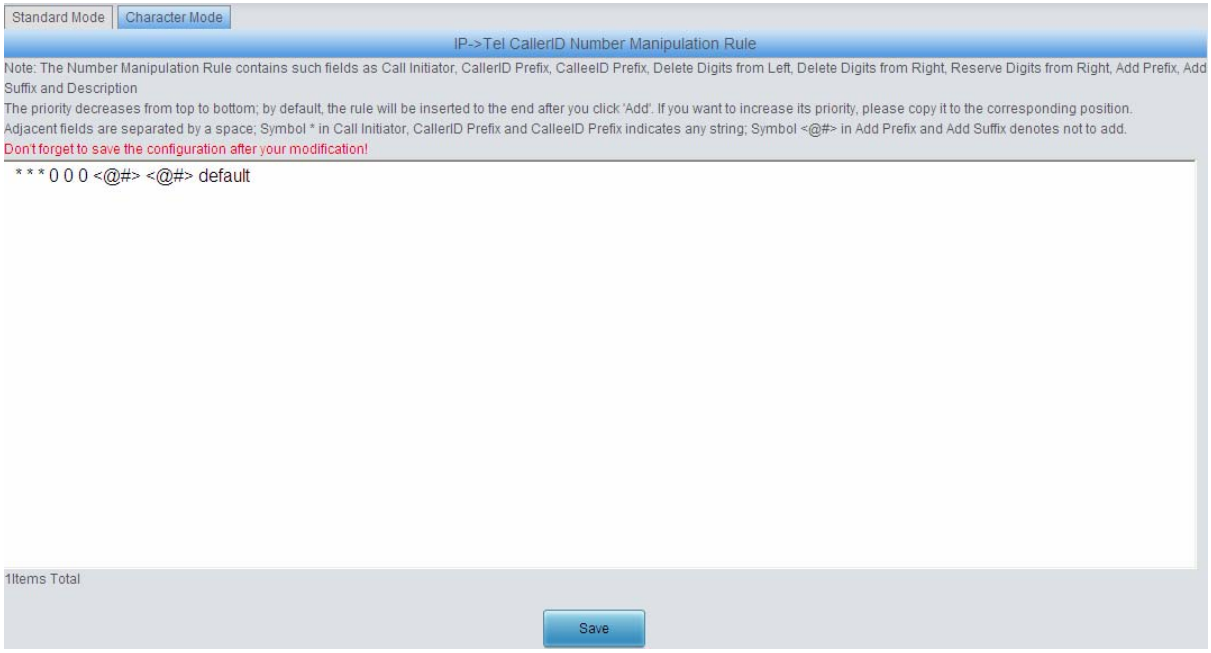
Standard Mode
Character Mode
IP->Tel CallerID Number Manipulation Rule
Note: The Number Manipulation Rule contains such fields as Call Initiator, CallerID Prefix, CalleeID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Prefix, Add Suffix and Description
The priority decreases from top to bottom; by default, the rule will be inserted to the end after you click 'Add'. If you want to increase its priority, please copy it to the corresponding position.
Adjacent fields are separated by a space; Symbol * in Call Initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol <@#> in Add Prefix and Add Suffix denotes not to add.
Don't forget to save the configuration after your modification!
* * * 0 0 0 <@#> <@#> default
1Items Total
Save

Figure 3-72 IP→Tel CallerID Manipulation Interface (Character)

3.9.2 IP to Tel CalleeID

The number manipulation process for IP→Tel CalleeID is almost the same as that for IP→Tel CallerID; only the number to be manipulated changes from CallerID to CalleeID. See Figure 3-73, Figure 3-74 for IP→Tel CalleeID Manipulation interface. The configuration items on this interface are the same as those on ***IP→Tel CallerID Manipulation Interface*** (Figure 3-69).

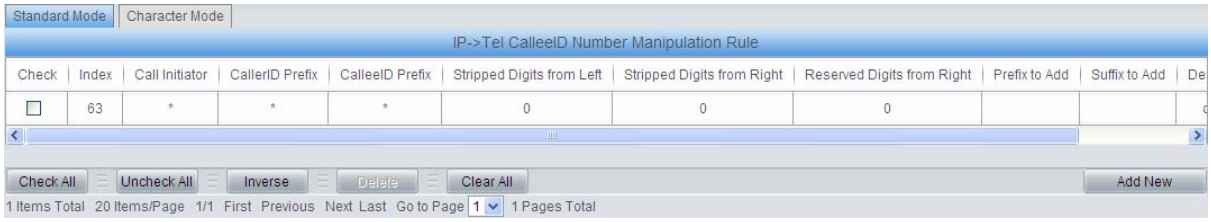

Figure 3-73 IP→Tel CalleeID Manipulation Interface(Standard)

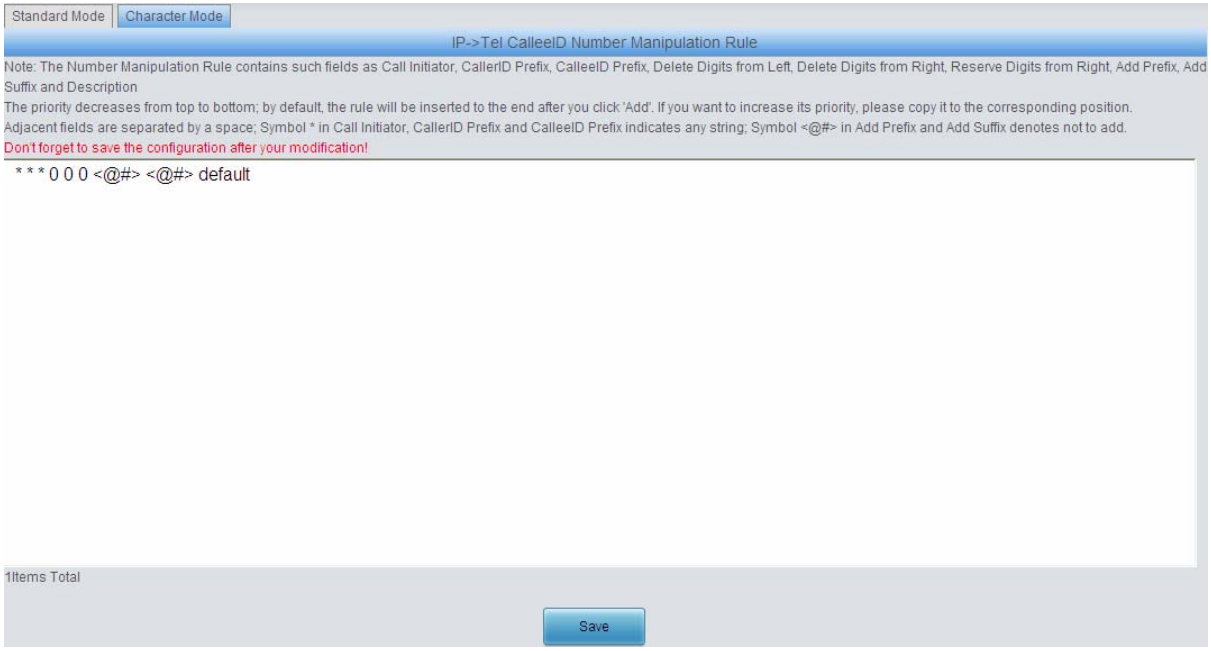
Standard Mode
Character Mode
IP->Tel CalleeID Number Manipulation Rule
Note: The Number Manipulation Rule contains such fields as Call Initiator, CallerID Prefix, CalleeID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Prefix, Add Suffix and Description
The priority decreases from top to bottom; by default, the rule will be inserted to the end after you click 'Add'. If you want to increase its priority, please copy it to the corresponding position.
Adjacent fields are separated by a space; Symbol * in Call Initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol <@#> in Add Prefix and Add Suffix denotes not to add.
Don't forget to save the configuration after your modification!
* * * 0 0 0 <@#> <@#> default
1Items Total
Save

Figure 3-74 IP→Tel CalleeID Manipulation Interface (Character)

3.9.3 Tel to IP CallerID

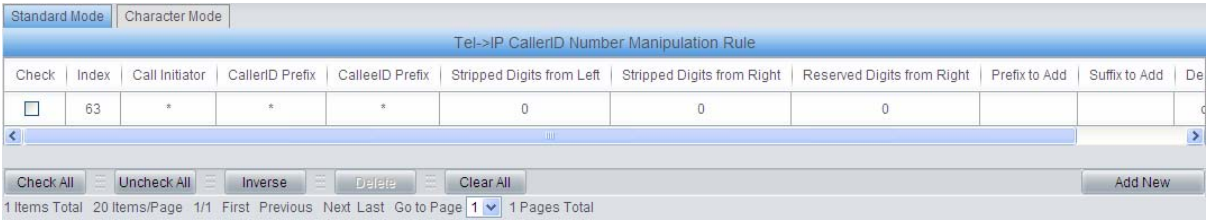

Figure 3-75 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-75 for the Tel→IP CallerID manipulation interface under the Standard mode. A new number manipulation rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-76 for the Tel→IP CallerID manipulation rule adding interface. You may use the default values of all the other configuration items herein.

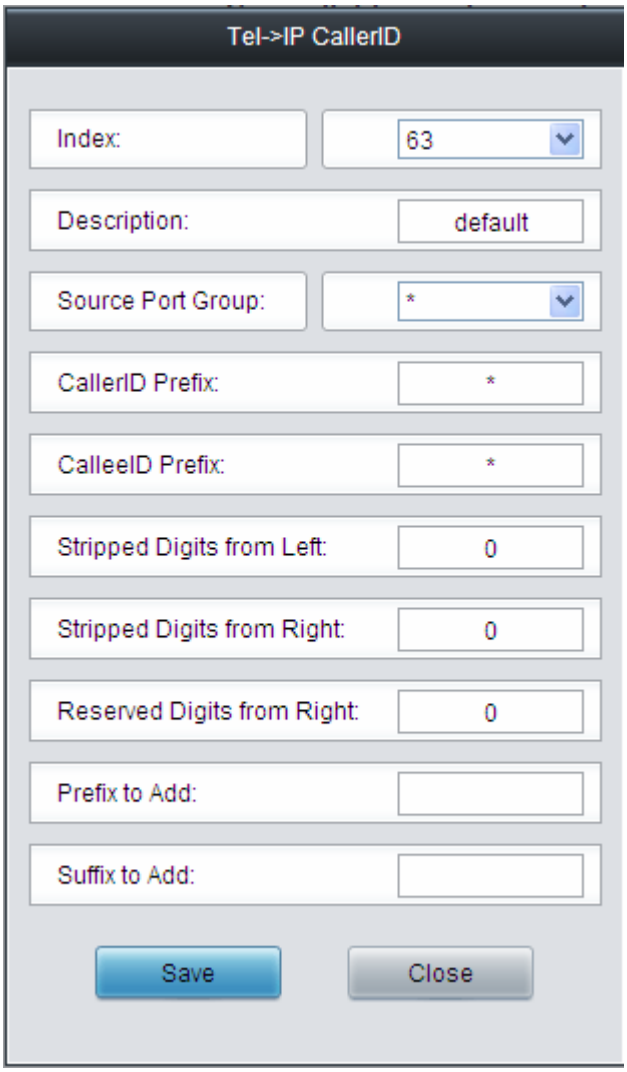

Figure 3-76 Add Tel→IP CallerID Manipulation Rule

The table below explains the items shown in the above figure.

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call

	matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule, with the default value of *default*.
Source Port Group (Call Initiator)	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '**-**' and '**,**'. ('**-**' is used between two characters to indicate any character between these two characters. '**,**' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Call Initiator*** specify a number manipulation rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

Note: The number manipulation is performed in 5 steps by the order of the following configuration items: *Stripped Digits from Left*, *Stripped Digits from Right*, *Reserved Digits from Right*, *Prefix to Add* and *Suffix to Add*.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

Click ***Modify*** in Figure 3-75 to modify a number manipulation rule. See Figure 3-77 for the Tel→IP CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the ***Add Tel→IP CallerID Manipulation Rule*** interface. Note that the item ***Index*** cannot be modified.

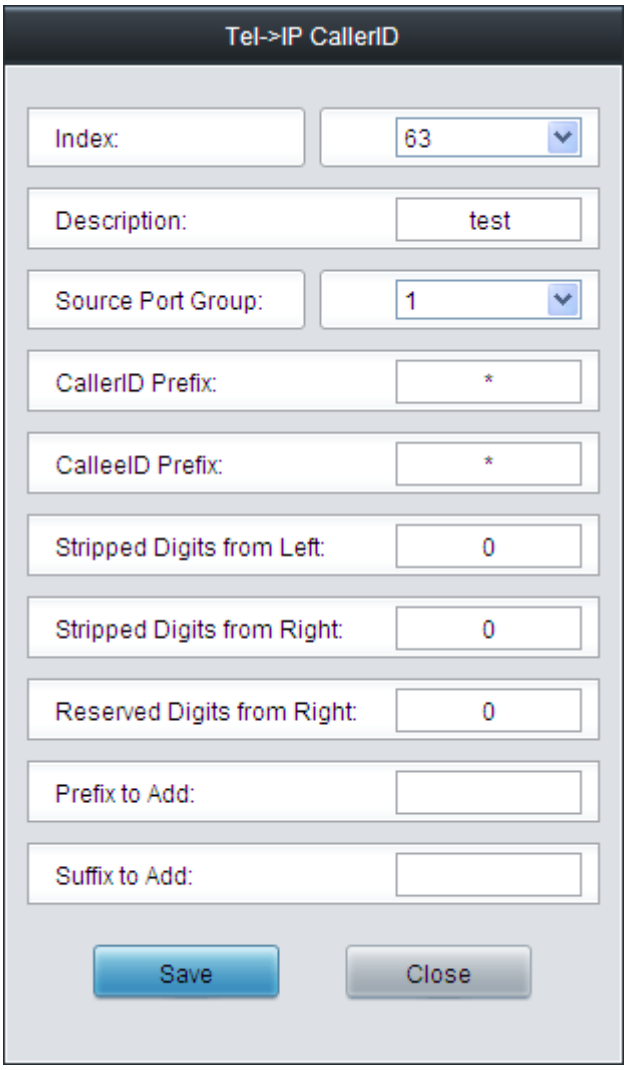

Figure 3-77 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-75 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all number manipulation rules at a time, click the ***Clear All*** button in Figure 3-75.

See Figure 3-78 for the Tel→IP CallerID Manipulation Interface under the Character mode. You can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

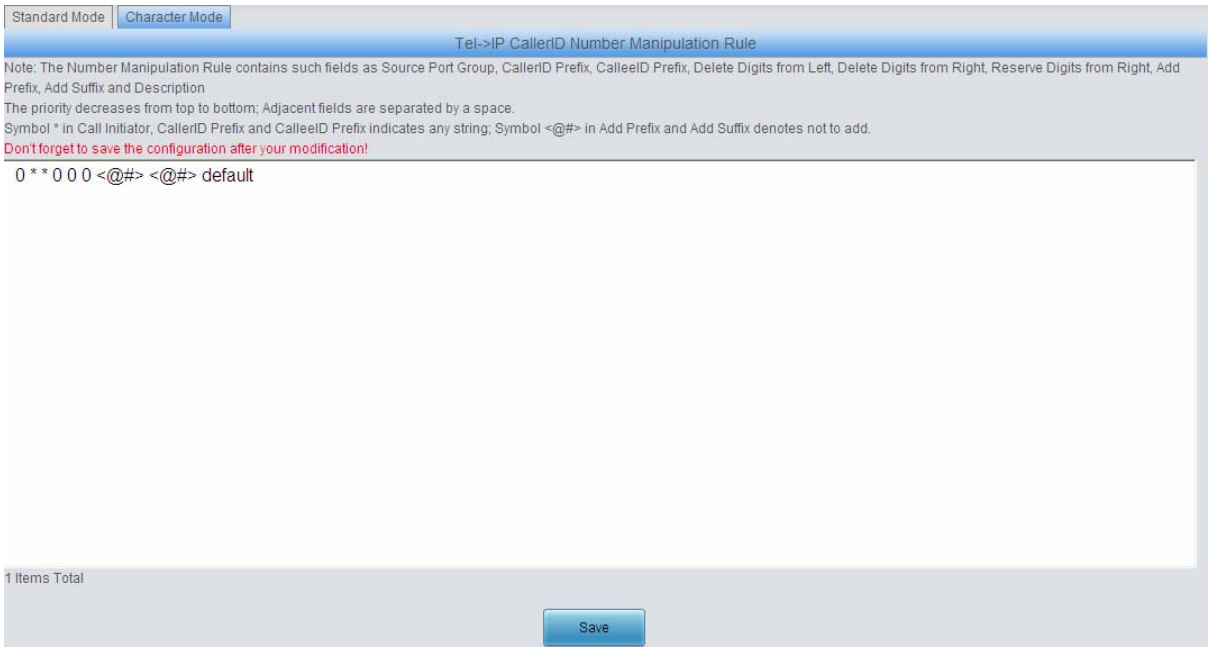
Standard Mode
Character Mode
Tel->IP CallerID Number Manipulation Rule
Note: The Number Manipulation Rule contains such fields as Source Port Group, CallerID Prefix, CalleeID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Prefix, Add Suffix and Description
The priority decreases from top to bottom; Adjacent fields are separated by a space.
Symbol * in Call Initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol <@#> in Add Prefix and Add Suffix denotes not to add.
Don't forget to save the configuration after your modification!
0 * * 0 0 0 <@#> <@#> default
1 Items Total
Save

Figure 3-78 Tel→IP CallerID Manipulation Interface (Character)

3.9.4 Tel to IP CalleeID

The number manipulation process for Tel→IP CalleeID is almost the same as that for Tel→IP CallerID; only the number to be manipulated changes from CallerID to CalleeID. See Figure 3-79, Figure 3-80 for the Tel→IP CalleeID manipulation interface. The configuration items on this interface are the same as those on ***Tel→IP CallerID Manipulation Interface*** (Figure 3-75).

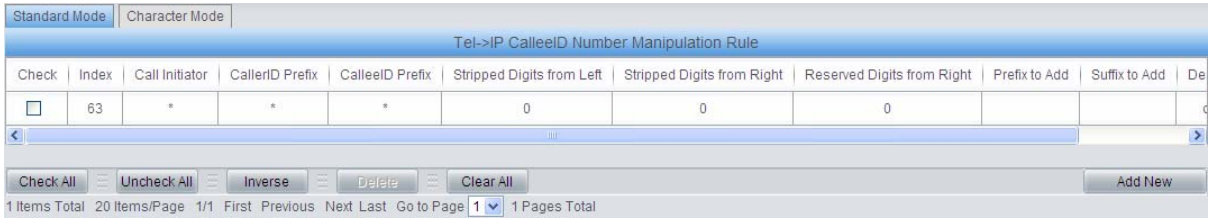

Figure 3-79 Tel→IP CalleeID Manipulation Interface (Standard)

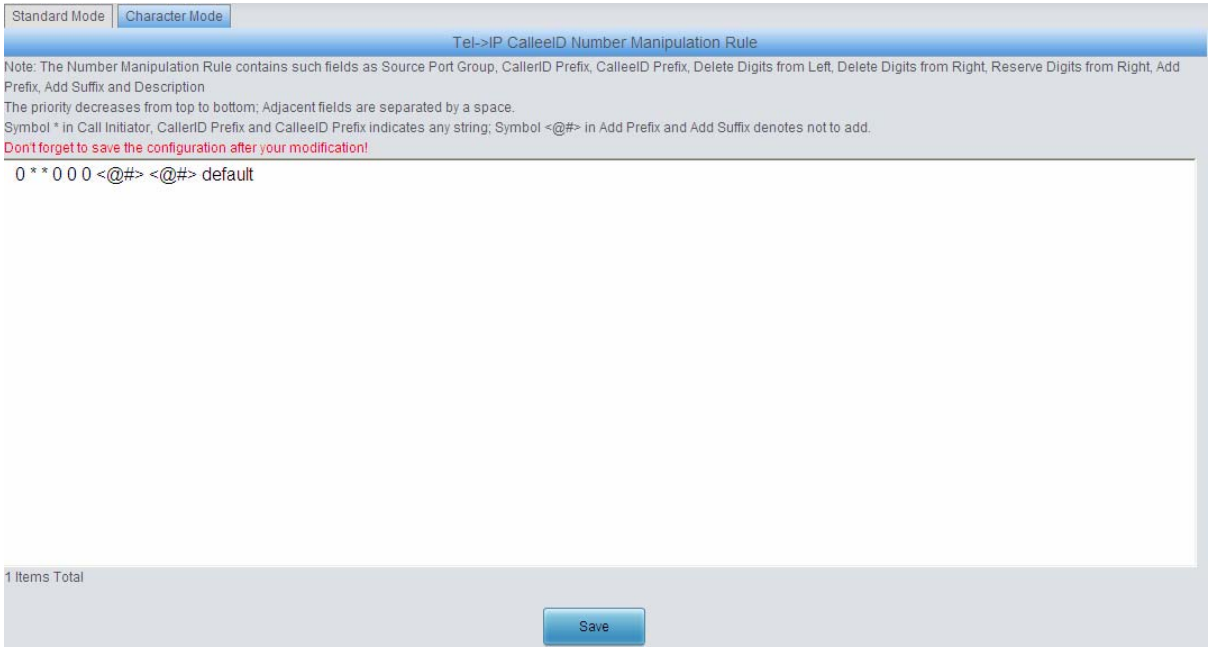
Standard Mode
Character Mode
Tel->IP CalleeID Number Manipulation Rule
Note: The Number Manipulation Rule contains such fields as Source Port Group, CallerID Prefix, CalleeID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Prefix, Add Suffix and Description
The priority decreases from top to bottom; Adjacent fields are separated by a space.
Symbol * in Call Initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol <@#> in Add Prefix and Add Suffix denotes not to add.
Don't forget to save the configuration after your modification!
0 * * 0 0 0 <@#> <@#> default
1 Items Total
Save

Figure 3-80 Tel→IP CalleeID Manipulation Interface (Character)

3.10 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, data backup and connectivity check. See Figure 3-81 for details.

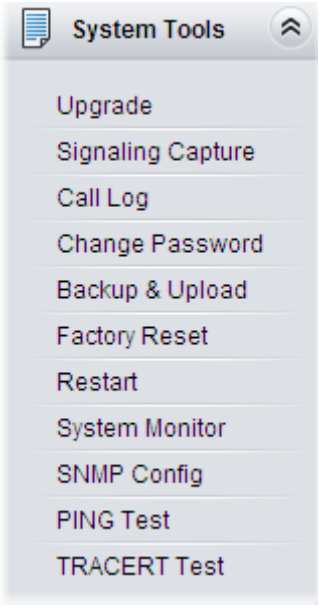

Figure 3-81 System Tools

3.10.1 Upgrade

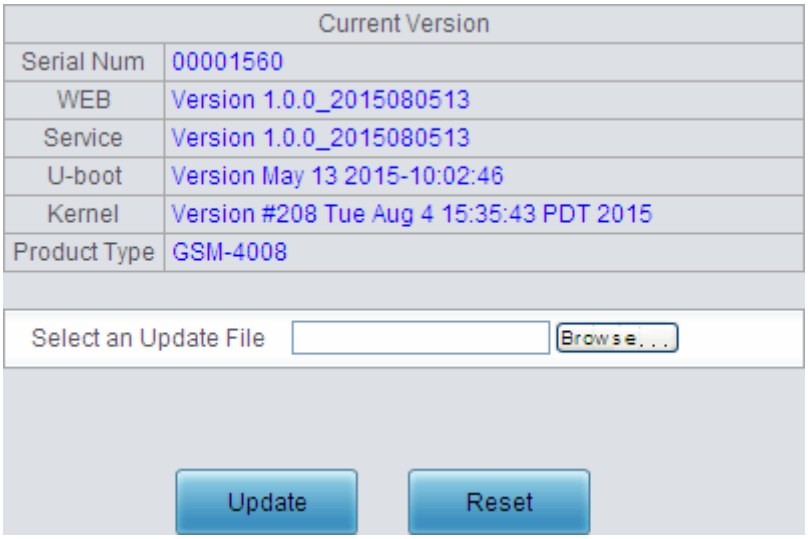

Figure 3-82 Upgrade Interface

See Figure 3-82 for the upgrade interface where you can upgrade the WEB, gateway service, kernel and firmware to new versions. Select the upgrade package "*.tar.gz" (The gateway will do MD5 verification before upgrading and will not start to upgrade until it passes the verification.) via ***Browse…*** and click ***Update***. Then the file uploading interface will appear. See Figure 3-83.

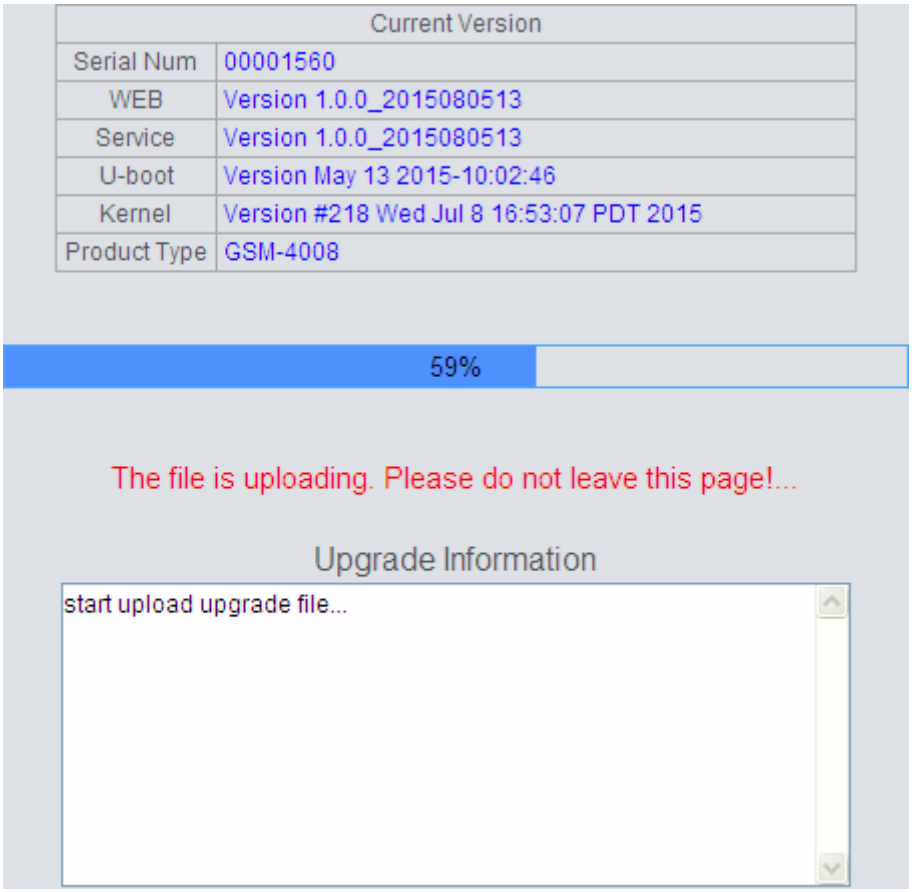

Figure 3-83 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-84 and you can learn the detailed upgrading information from the upgrade information box at the bottom.

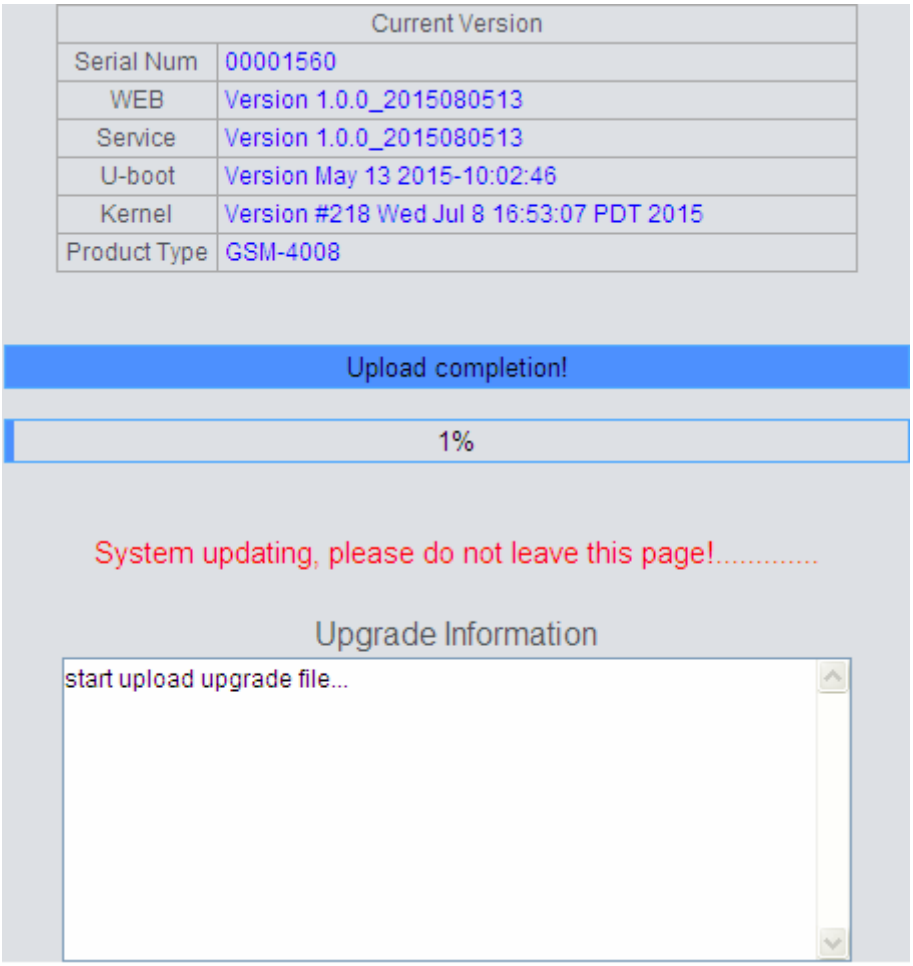

Figure 3-84 System Upgrading Interface

Note that clicking ***Reset*** can only delete the selected update file but not cancel the operation of ***Update***.

Note: Please contact our technicians if you need to downgrade the gateway to an old version. An improper operation may cause unexpected problems.

3.10.2 Signaling Capture

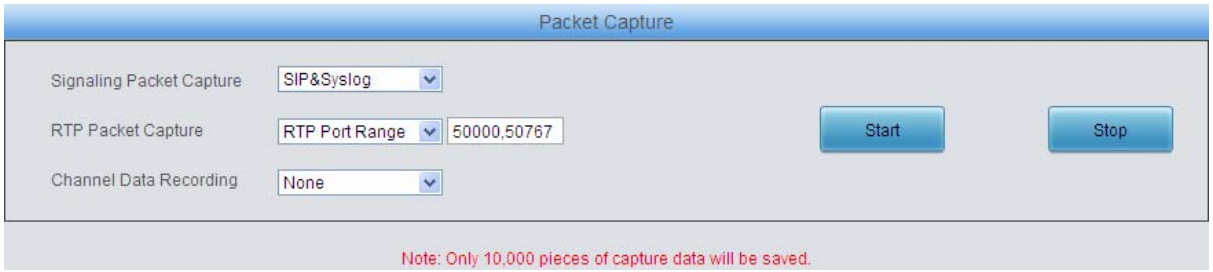

Figure 3-85 Signaling Capture Interface

See Figure 3-85 for the Signaling Capture interface. Packet capture contains Signaling Packet Capture, RTP Packet Capture and Channel Data Recording. You can select either of them to start the capture according to your requirement. Click ***Start*** to start capturing packets. Click ***Stop*** to stop the capture and download the captured packets.

3.10.3 Call Log

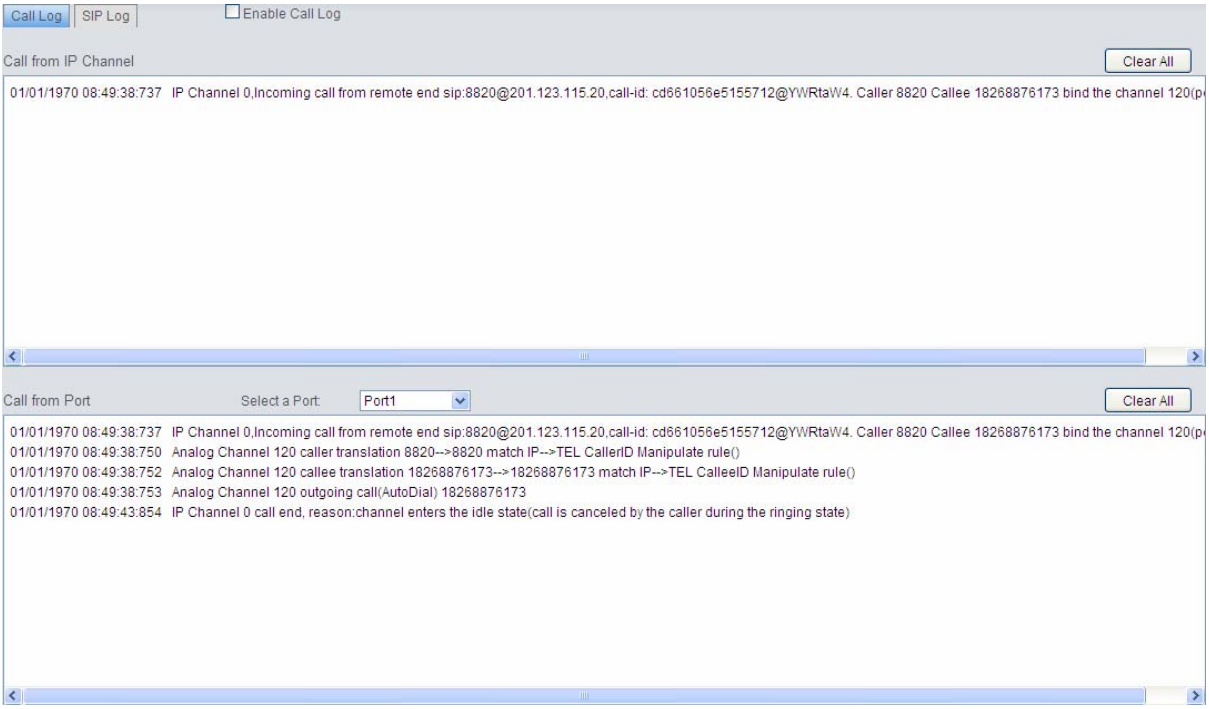
Call Log
SIP Log
Enable Call Log
Call from IP Channel
Clear All
01/01/1970 08:49:38:737 IP Channel 0,Incoming call from remote end sip:8820@201.123.115.20,call-id: cd661056e5155712@YWRtaW4. Caller 8820 Callee 18268876173 bind the channel 120(p
Call from Port
Select a Port:
Port1
Clear All
01/01/1970 08:49:38:737 IP Channel 0,Incoming call from remote end sip:8820@201.123.115.20,call-id: cd661056e5155712@YWRtaW4. Caller 8820 Callee 18268876173 bind the channel 120(p
01/01/1970 08:49:38:750 Analog Channel 120 caller translation 8820-->8820 match IP-->TEL CallerID Manipulate rule()
01/01/1970 08:49:38:752 Analog Channel 120 callee translation 18268876173-->18268876173 match IP-->TEL CalleeID Manipulate rule()
01/01/1970 08:49:38:753 Analog Channel 120 outgoing call(AutoDial) 18268876173
01/01/1970 08:49:43:854 IP Channel 0 call end, reason:channel enters the idle state(call is canceled by the caller during the ringing state)

Figure 3-86 Call Log Interface

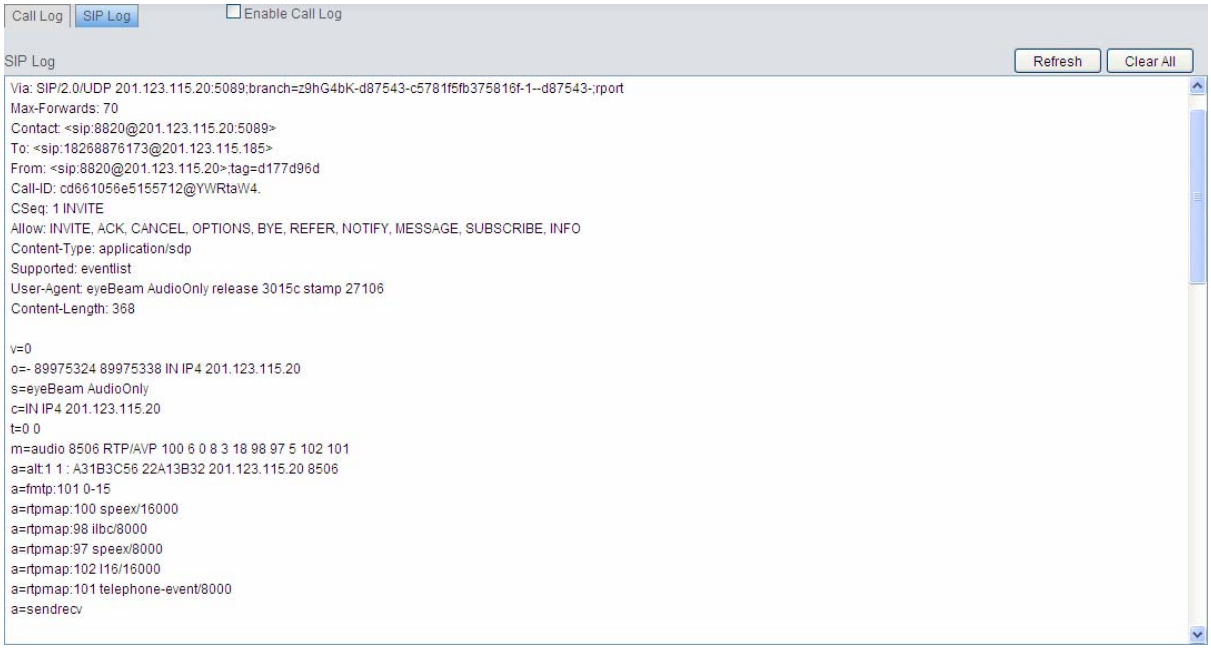

Figure 3-87 SIP Log Interface

See Figure 3-86, Figure 3-87 for the Call Log interface. Click the checkbox before ***Enable Call Log*** to enable the call log feature, including ***Call Log*** and ***SIP Log***. ***Call from IP Channel*** displays the call log information generated on all IP channels, and ***Call from Port*** displays the call log information generated on the port you select. All the SIP related information will be displayed in ***SIP Log***.

3.10.4 Change Password

Figure 3-88 Password Changing Interface

See Figure 3-88 for the password changing interface where you can change username and password of the gateway. Enter the current password, the new username and password, and then confirm the new password. After configuration, click ***Save*** to apply the new username and password or click ***Reset*** to restore the configurations. After changing the username and password, you are required to log in again.

3.10.5 Backup & Upload

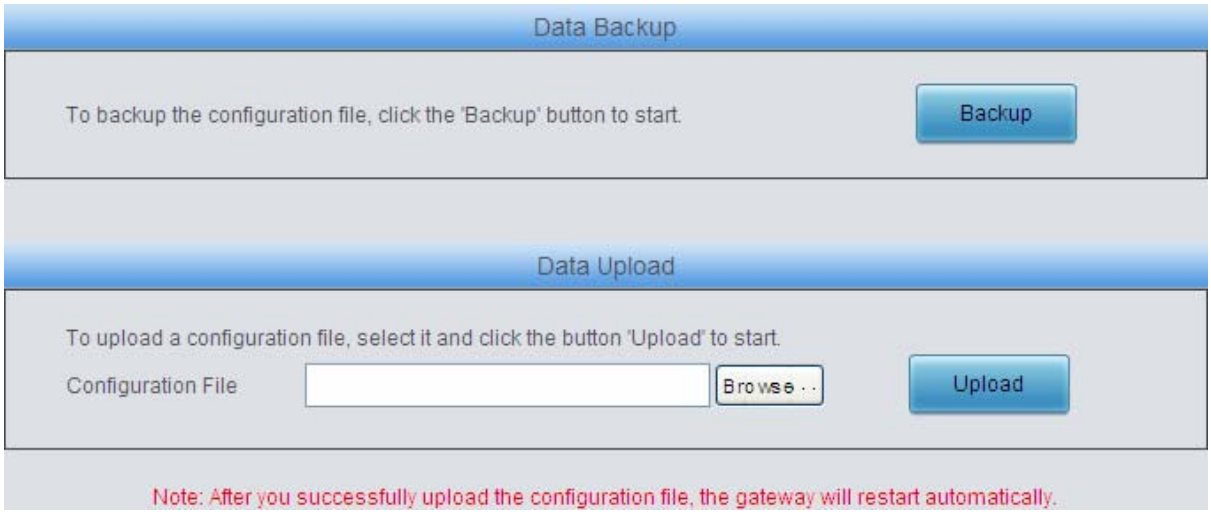
Data Backup
To backup the configuration file, click the 'Backup' button to start.
Backup
Data Upload
To upload a configuration file, select it and click the button 'Upload' to start.
Configuration File
Browse...
Upload
Note: After you successfully upload the configuration file, the gateway will restart automatically.

Figure 3-89 Backup & Upload Interface

See Figure 3-89 for the backup and upload interface. To back up the configuration file to your PC, just click ***Backup***. To upload a configuration file, select it via ***Browse…*** and click ***Upload***.

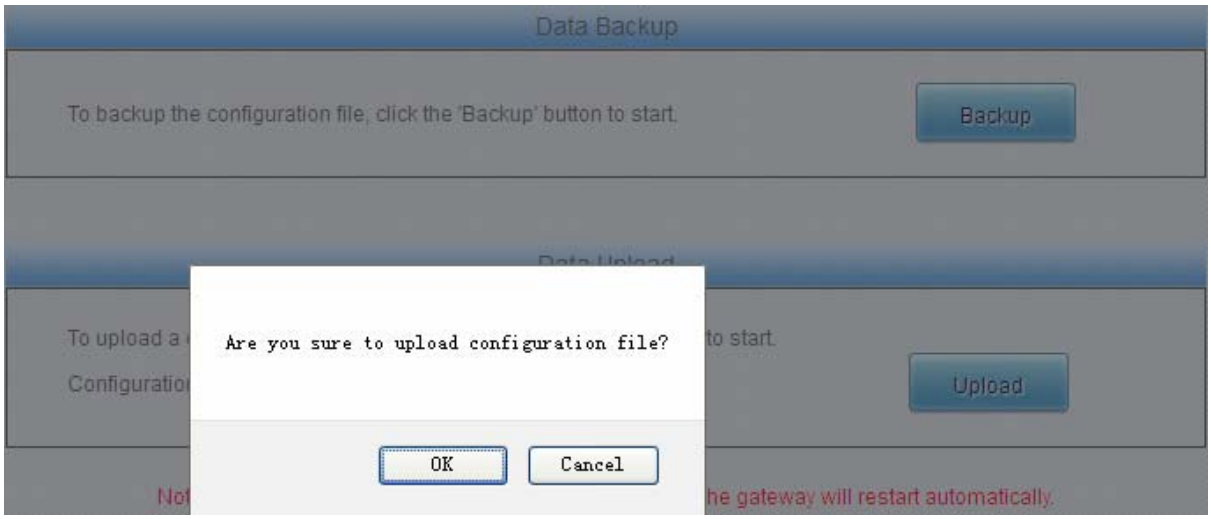

Figure 3-90 Backup & Upload & Prompt Interface

Click ***OK*** on the prompt box (Figure 3-90) to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. See Figure 3-91. The gateway will overwrite the current configurations with the uploaded data after restart. Click ***Cancel*** to cancel this upload directly.

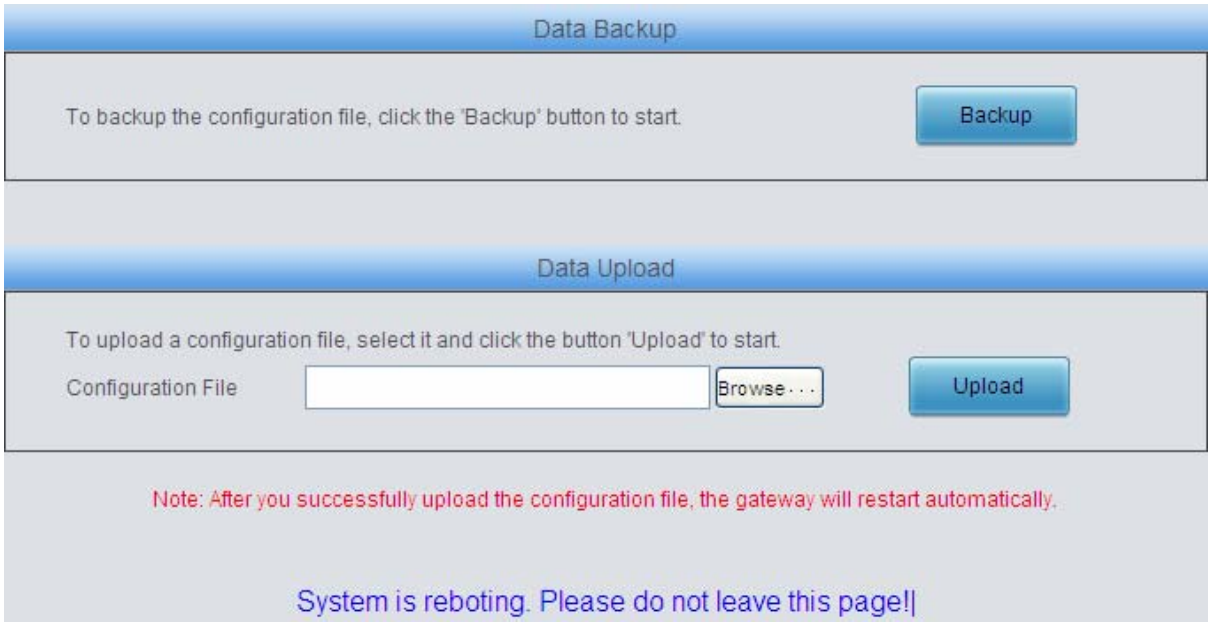
Data Backup
To backup the configuration file, click the 'Backup' button to start.
Backup
Data Upload
To upload a configuration file, select it and click the button 'Upload' to start.
Configuration File
Browse...
Upload
Note: After you successfully upload the configuration file, the gateway will restart automatically.
System is reboting. Please do not leave this page!

Figure 3-91 Configuration File Uploading Interface

3.10.6 Factory Reset

Factory Reset
Click the button 'Reset' below to restore to factory settings.
Reset
Note: After you successfully restore the gateway to factory settings, the gateway will restart automatically and its IP address will be restored to the default one.

Figure 3-92 Factory Reset Interface

See Figure 3-92 for the factory reset interface. Click ***Reset*** to restore all configurations on the gateway to factory settings.

3.10.7 Restart

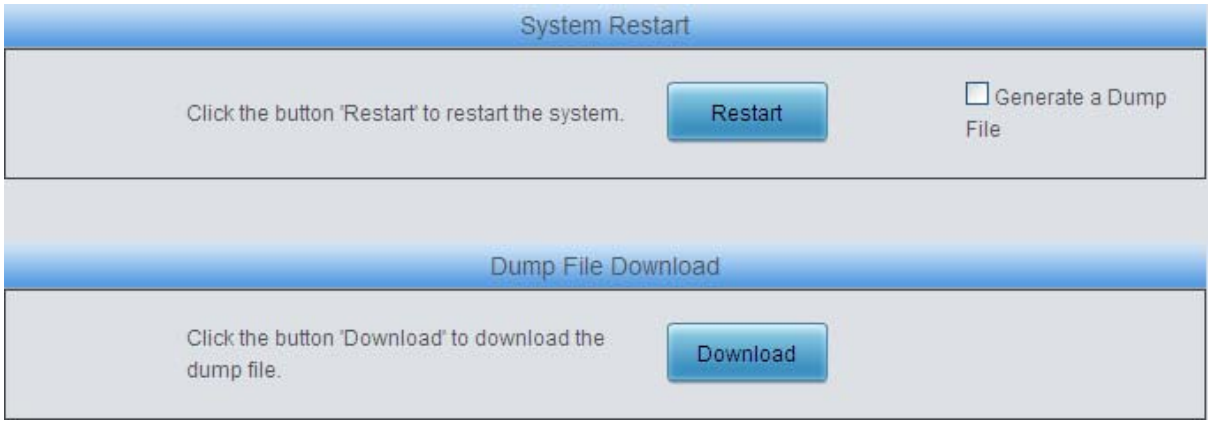
System Restart
Click the button 'Restart' to restart the system.
Restart
Generate a Dump File
Dump File Download
Click the button 'Download' to download the dump file.
Download

Figure 3-93 System Restart Interface

See Figure 3-93 for the restart interface. Click ***Restart*** under the service restart interface to restart the gateway service or click ***Restart*** under the system restart interface to restart the whole gateway system. A dump file will be generated each time you restart the service or the system. Click ***Download*** and you can download it to help troubleshoot issues.

3.10.8 System Monitor

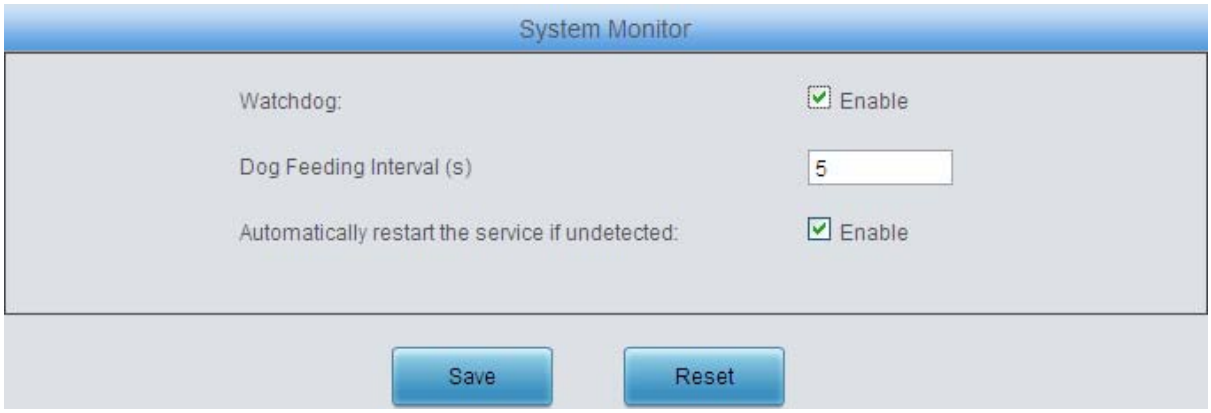

Figure 3-94 System Monitor Configuration Interface

See Figure 3-94 for the System Monitor Configuration interface. Watchdog is a timing reset system used to avoid application crash. You can set the dog feeding interval when this feature is enabled. The feeding interval is calculated by s, with the value range of 1~15s. By default, this feature is enabled with the default value of 5s. As the feature 'Automatically restart the service if undetected' is enabled, the service application will restart automatically if it is not detected by the gateway guard application. By default, this feature is enabled.

3.10.9 SNMP Config

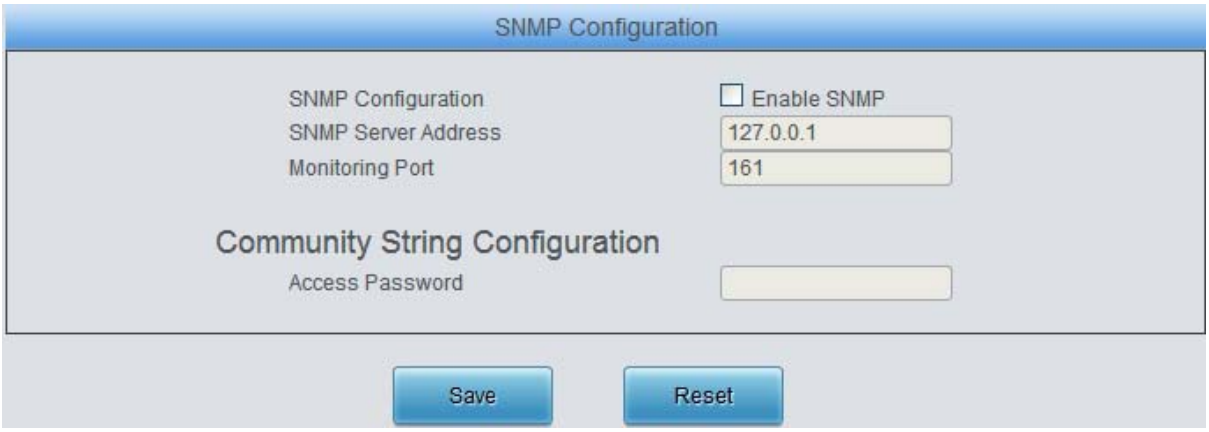

Figure 3-95 SNMP Configuration Interface

See Figure 3-95 for the SNMP configuration interface. If the SNMP feature is enabled, once the gateway receives a request from the SNMP management software, it will collect relevant information and reply them to the SNMP management software. By default, the SNMP feature is disabled. The available information includes kernel version, CPU usage, processes, memory usage, startup information, LAN status and etc. Currently, the gateway only provides the community string for information acquisition. The table below explains the configuration items shown in Figure 3-95.

Item	Description
SNMP Server Address	IP address of SNMP.
Monitoring Port	Monitoring Port for SNMP on the gateway.
Access Password	Community string used for information acquisition.

3.10.10 PING Test

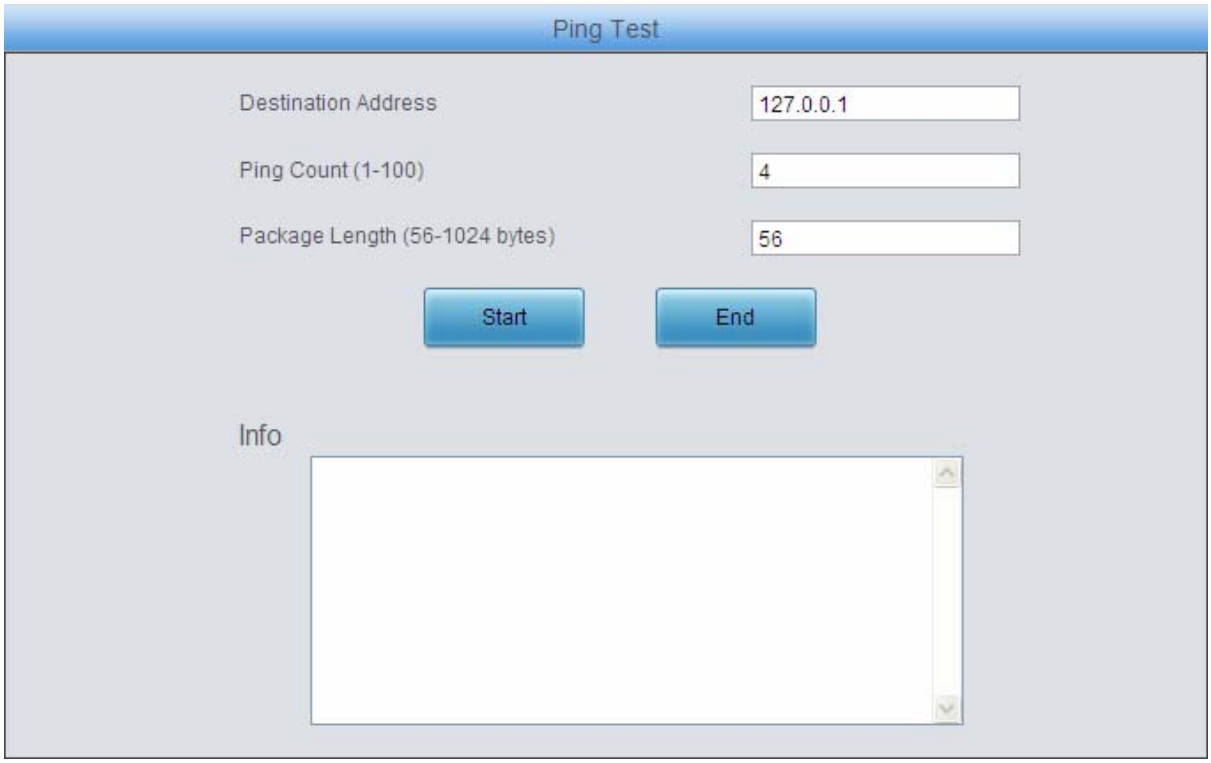

Figure 3-96 Ping Test Interface

See Figure 3-96 for the Ping test interface. A Ping test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items shown in the above figure.

Item	Description
Destination Address	Destination IP address or domain name on which the Ping test is executed.
Ping Count	The number of times that the Ping test should be executed. Range of value: 1~100.
Package Length	Length of the data package used in the Ping test. Range of value: 56~1024 bytes.
Info	The information returned during the Ping test, helping you to learn the network connection status between the gateway and the destination address.

After configuration, click ***Start*** to execute the Ping test; click ***End*** to terminate it immediately.

3.10.11 TRACERT Test

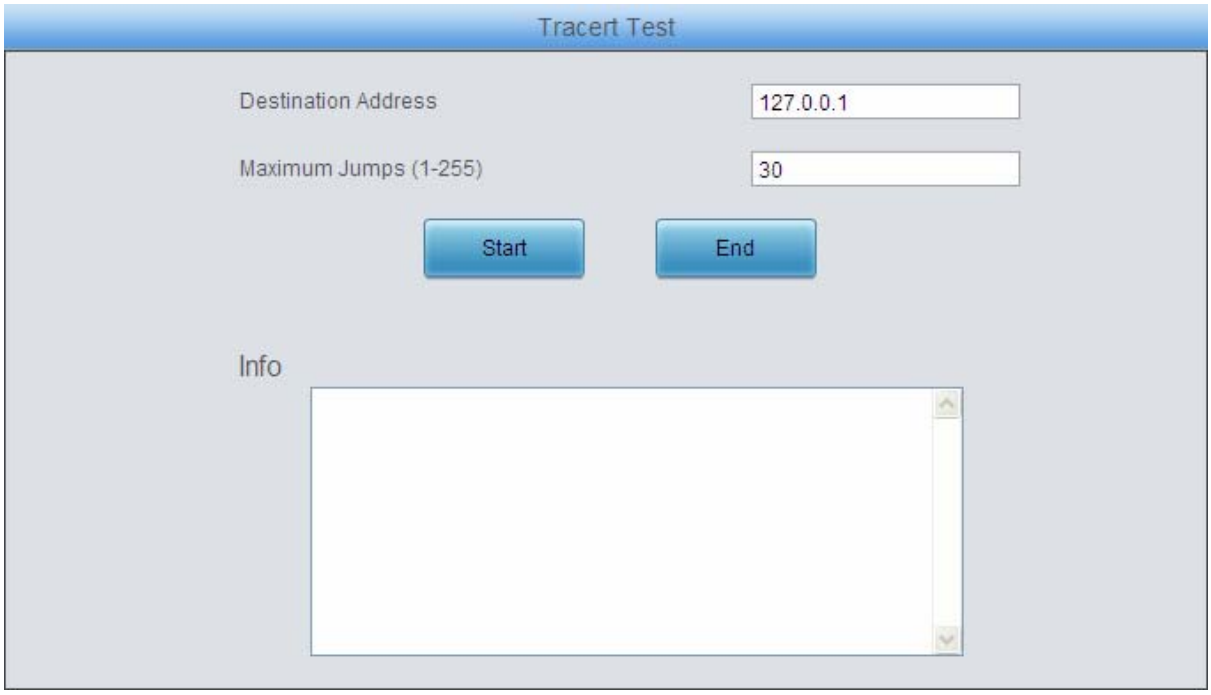

Figure 3-97 Tracert Test Interface

See Figure 3-97 for the Tracert test interface. A Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items shown in the above figure.

Item	Description
Source IP Address	Source IP address where the Tracert test is initiated.
Destination Address	Destination IP address on which the Tracert test is executed.
Maximum Jumps	Maximum number of jumps between the gateway and the destination address which are returned by the Tracert test. Range of value: 1~255.
Info	The information returned during the Tracert test, helping you to learn the detailed information about the jumps between the gateway and the destination address.

After configuration, click ***Start*** to execute the Tracert test; click ***End*** to terminate it immediately.

Appendix A Technical Specifications

Dimensions

260×153×30 mm³

Weight

Net: 1.2 kg

Gross: 1.8 kg

Environment

Operating temperature: 0℃—55℃

Storage temperature: -20℃—85℃

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

LAN

Amount: 2 (10/100 BASE-TX (RJ-45))

Self-adaptive bandwidth supported

Auto MDI/MDIX supported

Console Port

Amount: 1 (RS-232)

Baud rate: 115200bps

Connector: RJ45 to DB-9 Connector

Data bits: 8 bits

Stop bit: 1 bit

Parity unsupported

Flow control unsupported

Note: Follow the above settings to configure the serial port; or it may work abnormally.

Power Requirements

Input power: 12V DC ±10%

Input Current: ≥3A DC

Signaling & Protocol

SIP signaling

Supported protocol: SIP V1.0/2.0, RFC3261

Network Protocol

IP v4, UDP/TCP, PPPoE, DHCP,

FTP/TFTP ARP, RARP, NTP,

HTTP, Telnet

Audio Encoding & Decoding

G.711A	*64 kbps*
G.711U	*64 kbps*
G.729A/B	*8 kbps*
G723	*5.3/6.3 kbps*
G722	*64 kbps*
AMR	*4.75 kbps*
iLBC	*13.3/15.2 kbps*

Sampling Rate

8kHz

Wireless Feature

GSM Frequency band: 850/900/1800/1900MHz

SMS CODEC: ASCII/UCS2

Safety

Lightning resistance: Level 4

Appendix B Troubleshooting

Q1. What to do if I forget the IP address of the GSM gateway?

There are two ways to get the IP address:

1) Long press the Reset button on the gateway to restore to factory settings. The default IP address is 192.168.1.101
2) Make a call to any wireless port and press the function key to query the IP address. See 3.5.5 Function Key for more details.

Q2. In what cases can I conclude that the SMG-B gateway is abnormal and turn to Synway's technicians for help?

a) During runtime, the run indicator does not flash or the alarm indicator lights up or flashes, and such error still exists even after you restart the device or restore it to factory settings.
b) Voice problems occur during call conversation, such as that one party or both parties cannot hear the voice or the voice quality is unacceptable.
c) The port of the gateway is well connected, but the channel indicator never lights up after the gateway startup or the color it lights up does not comply with the actual state or port type.

Other problems such as inaccessible calls, failed registrations, incorrect numbers are probably caused by configuration errors. We suggest you refer to Chapter 3 WEB Configuration for further examination. If you still cannot figure out or solve your problems, please feel free to contact our technicians.

Q3. What to do if I cannot enter the WEB interface of the GSM gateway after login?

This problem may happen on some browsers. To settle it, follow the instructions here to configure your browser. Enter 'Tools > Internet Options >Security Tab', and add the current IP address of the gateway into 'Trusted Sites'. If you changes the IP address of the gateway, add your new IP address into the above settings too.

Q4. Is there any cell-phone APP can make calls to the GSM gateway?

Yes. Linphone is a soft SIP phone that is supported by multiple platforms, such as Linux, Windows, iOS, Android, etc. It must be registered to the SIP registrar server before dialing to other SIP devices or PSTN telephones,

Q5. Which RTP codecs are supported by the GSM gateway?

At present, the supported RTP codecs are: G.711A, G.711u, G.729, G.723, G.722, AMR and iLBC.

Appendix C Technical/sales Support

Thank you for choosing Synway. Please contact us should you have any inquiry regarding our products. We shall do our best to help you.

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